THE JOURNAL
OF THE
ACOUSTICAL SOCIETY OF AMERICA

Volume 11 Number 4

APRIL • 1940

Some Practical Aspects of Architectural Acoustics*

VERN O. KNUDSEN
University of California at Los Angeles, Los Angeles, California
(Received December 10, 1939.)

THE recent work[1] of Morse, Hunt, Bolt, and others on the vibration of sound in rooms provides a most promising approach to the solutions of two problems of prime importance in architectural design: (1) the optimal form or shape of all rooms in which good acoustics is a significant factor, and (2) the choice and distribution of absorptive and reflective materials for the interior surfaces of such rooms.

Experience and empiricism, supported by certain deductions based on the assumption that sound waves travel in straight lines and are reflected in the same manner as are nonspinning balls on a billiard table, have been our guide in designing the shapes of rooms. But the assumption that sound waves travel in a room as do balls on a billiard table, or as do rays of light in a room bounded by mirrors, is only approximately valid for high frequencies—above about 1000 cycles—and is seriously false for low frequencies—below about 200 cycles. Sound is a wave motion; the wave-length of many sounds which are used in rooms is not small compared with the dimensions of the rooms; therefore we are obliged to solve the problems of room acoustics by the principles of *physical acoustics*, which regards sound as a wave phenomenon, and not by the principles of *geometrical acoustics*, which, though very helpful in guiding the acoustical design of large rooms, may lead to serious errors, especially in small rooms.

Every room, except those enclosed by totally absorptive boundaries, is a resonant chamber, capable of being excited into resonant vibration of one or many of a triply-infinite series of frequencies—these frequencies being determined by the dimensions of the room and the velocity of sound. In a rectangular room, 8 feet by 8 feet by 9.5 feet (high), a room in which the writer has conducted many acoustical experiments,[2] the resonant or normal modes of vibration, at a room temperature of 70°F, have frequencies of 59.1, 70.3, 91.7, 99.6, 115.9, ··· cycles per second. There are 27 such normal modes with frequencies lower than 200 cycles, and several hundred modes with higher frequencies. Whenever sound is produced in such a room, at least

* Based on a paper presented at the Symposium on Architectural Acoustics at the joint meeting of the Acoustical Society of America and the Iowa Chapter of the American Institute of Architects, Iowa City, November 3, 1939.

[1] See especially the J. Acous. Soc. Am., July, 1939, for references to this work.

[2] "Resonance in Small Rooms," J. Acous. Soc. Am. **4** 20–37 (1932).

one, and usually many, of these resonant vibrations are excited. The resulting sound pattern in the room is a composite of the sound radiated directly by the source and the resonant vibrations excited by the source of sound. If, in the above room, the sound source have a frequency of 59.1 cycles, or 70.3 cycles, or any one of the resonant or normal frequencies, that frequency will be strongly excited, and the room is made to *sing* or resonate at this frequency.

The would-be baritone singing in the bath plays on these resonant frequencies; he is in fact using the bathroom as a musical instrument. The rich quality of his sustained tones may be attributes of the room rather than the excellence of his voice. A small, resonant room having dimensions in the ratio of 4 : 5 : 6 would provide an interesting setting for a male quartet; the familiar *do mi sol do*, based on the major triad, could be pitched to coincide exactly with certain resonant frequencies of the room, thus imparting an extraordinary power and quality to such chords.

In every room, the space pattern of the intensity of sustained sounds and the growth and decay of transient sounds throughout the room are largely determined by these normal modes of vibration; and, while the frequencies of these normal modes are determined primarily by the dimensions and shape of the room, their intensities and rates of decay are largely influenced by the distribution of absorptive materials over the interior boundaries of the room. The proper control of the distribution of sound energy throughout a room and of the growth and decay of transient sounds are the prime objectives of good acoustical design. We shall be able to attain scientifically rigorous and predeterminable control of these objectives only by a detailed and comprehensive study of the normal modes of vibration in a room. This does not appear to be a simple problem, as those of us who have followed recent developments in room acoustics well know. Fortunately, for most speech rooms and for practically all commercial rooms, the approximate reverberation formulas we have been using since the days of W. C. Sabine, based on geometrical acoustics, are quite satisfactory if used with a proper understanding of the behavior of sound in rooms. But the scientific design of the ideal music room should be based on the more rigorous concepts of physical acoustics, now in process of evolution.

There are other acoustical characteristics of small rooms which are not anticipated by the older geometrical acoustics but are readily predicted by the modern physical acoustics. Referring again to the small room, 8 feet by 8 feet by 9.5 feet, a tone of 70.3 cycles per second will set the room into a resonant vibration of this frequency. The tone will be pure and will be sustained at a relatively high intensity. The reverberation or decay of such a tone will consist of a *continuation of the same pure tone*, and the decay will be *strictly logarithmic*. Tones of somewhat higher pitch, where the resonant frequencies in the room are not widely separated, for example between 90 and 116 cycles, will usually excite, besides the impressed or forced frequency, two or more contiguous resonant or normal modes; and the reverberation of such tones will consist not of the single-frequency impressed tone but of the simultaneous decay of the two or more normal vibrations.[3] In this type of reverberation the well-known phenomenon of beats occurs.

Since, in general, reverberation consists of the simultaneous decay of several resonant vibrations of nearly the same frequency, the pulsating or beating effects mentioned above are a characteristic of reverberation. These pulsations often have an effect upon reverberant sounds which is similar to that of the vibrato, an effect which may prove to be as important in determining the acoustical quality of music rooms, especially small ones, as is the reverberation time itself. Certainly the detailed nature of sound decay, especially during the early stages of decay, and not merely the average rate of decay, affects the acoustical excellence of music rooms. Standards of what is most desirable musically, as well as techniques for attaining these standards, must yet be worked out. Here is a field of research which not only is fascinating, but also holds promise of providing marked improvements in the acoustics of music rooms.

An appreciation of the theory of room resonance will save the architect and engineer from

[3] See Fig. 9, J. Acous. Soc. Am. **4**, 33 (1932).

errors in the placement of absorptive materials not anticipated by the older reverberation theories. For example, suppose the ceiling only of the small room to which we already have referred were treated with a highly absorptive material. According to the standard reverberation formulas, such treatment should provide this room with a reverberation time well below one second, and consequently with good acoustical quality. However, all sounds having a component frequency of about 70.3 cycles, or integral multiples of 70.3 cycles, will excite resonant vibrations which vibrate in a horizontal direction, and since these encounter the absorptive ceiling only at grazing incidence, these vibrations die away slowly, giving the room a "boomy" and reverberant quality—the measured time of reverberation, even at 512 cycles, is more than two times as long as the calculated time. Both physical acoustics and experience teach us that good acoustical quality can be attained in such a room only by distributing the absorptive material on the walls as well as the ceiling, a lesson many architects and even some acoustical engineers are slow to learn.

A music room should be so dimensioned, shaped, and treated as to support and enhance the tonal quality and the tonal blending of a wide variety of musical sounds. The experiments of Saunders,[4] of Meinel,[5] and others, indicate that the best violins appear to be those which are free from pronounced resonant peaks and which have a relatively uniform response throughout their entire frequency range. If this be true, there is equally good reason to believe that the best music rooms are those which similarly are characterized by the absence of prominent resonances and by the presence of a uniform frequency response. Resonant peaks are most prominent in small rooms, and occur particularly for low pitched tones. Their troublesome effects can be effectively eliminated by the use of materials especially absorptive at the low frequencies, applied to the walls as well as the ceiling of the room. Such materials as acoustical tiles with a diaphragmatic facing, as wood veneer or thin wood sheathing or even plaster on lath, furred out so as to leave an air space behind it of at least an inch or two, should be used to suppress low pitched room resonances and to provide a relatively flat reverberation characteristic. The best music studios today (excepting very large ones) appear to be those with a nearly flat reverberation characteristic, with a reverberation time of about one second for small rooms, and increasing to 1.5 or even 2.0 seconds for large symphony and choral halls.

[4] F. A. Saunders, J. Acous. Soc. Am. **9**, 81–98 (1937).
[5] Hermann Meinel, Zeits. f. tech. Physik **10**, 297–307 (1938).

Some Typical Errors in Room Acoustics

The notion persists among many architects and builders that if an acoustical material be specified and used in any room, even if limited to the ceiling, the acoustical requirements for that room have been satisfied. Although it usually is better to do this than to do nothing, such a procedure often leads to poor acoustics. The writer has recently inspected a number of rooms which were unsatisfactory acoustically, even though in many instances absorptive materials had been installed in the ceilings, and in some instances on the walls as well as in the ceilings. A summary of the acoustical findings in certain of these rooms will illustrate the need for giving careful attention to the several factors which determine the outcome in room acoustics.

1. In a number of school buildings recently examined, acoustical plaster had been applied to the ceilings of the recitation rooms but ordinary hardwall plaster to the walls. When the width and length of these rooms were at least two or three times the ceiling height, and when the applied acoustical plaster had coefficients of sound absorption of not less than 0.15 at 128 cycles and not less than 0.28 at 512 to 2048 cycles, the outcome in acoustics was acceptable although not so good as it should be. But in some of the rooms the acoustical plaster was found to have coefficients as low as 0.05 at 128 cycles and 0.08 at 512 cycles, which is little or no better than ordinary plaster, and in such rooms the outcome in acoustics was very poor. Now a good brand of acoustical plaster, applied to a thickness of not less than one-half inch by a skillful plasterer, under competent supervision, over scratch and brown coats on metal lath on

channel irons or wood lath on wood studs, can be relied on to have coefficients in excess of 0.15 at 128 cycles and 0.28 at 512 to 2048 cycles. But inferior brands of acoustical plaster, or even first rate ones improperly applied, may lead to disappointing results. Unfortunately, the results found in most of the school buildings to which we have just referred were disappointing indeed —so much so that the school board responsible for these buildings has practically tabooed the use of acoustical plaster. It is true that most of these class rooms would have been much better than they are if the ceilings had been treated with a highly absorptive acoustical tile. However, this should not be interpreted to mean that the use of acoustical plaster should be discontinued. Acoustical plaster has been used for the walls and ceilings of every lecture and recitation room, and nearly every office, in the buildings recently completed at the University of California at Los Angeles. There has not been a single failure. The results are uniformly good in all rooms. The reverberation time is about 0.6 to 0.8 second in the offices, and about 0.8 to 1.2 seconds in the recitation and lecture rooms. Further, with the absorptive material distributed over the walls as well as the ceilings, the rooms are free from the resonance of horizontal modes of vibration, a defect frequently encountered when only the ceiling, especially in a small office, is treated with absorptive material. In small rooms, the acoustical result obtained from a moderately absorptive material applied to the walls as well as the ceiling is superior to that obtained from a highly absorptive material applied only to the ceiling. On the other hand, in rooms having large horizontal dimensions compared with that of the ceiling height, such as large public offices, cafeterias, etc., the absorptive material should be applied to the ceiling; in fact, it will be more effective in reducing reverberation and noise when applied to the ceiling than it will be in any other position.

2. Some court rooms in a new building recently inspected exhibit a number of acoustical features, some good and others inexcusably poor. The rooms are without windows, and thus remarkably free from outside noise. Absorptive materials are used lavishly, adorning both ceilings and walls. The rooms contain too much rather than too little absorption, especially on the wall, floor, and ceiling surfaces near the bench, the witness box, and the litigants' tables—surfaces which should be reflective rather than absorptive. The absence of such reflective surfaces is clearly evident; whenever the speaker faces his audience he can be heard very well, but when his back is turned he can be heard only with great difficulty, if at all. The intensity or loudness of the speech is further diluted by an immoderately high ceiling, probably deemed necessary to reflect the high and dignified functions of the court. Finally, the noise from the air conditioning equipment is disturbingly loud—a defect which could be corrected by the insertion of suitable sound filters in the supply and exhaust ducts.

It is unfortunate indeed that so little attention is devoted to the acoustical design of our court and council chambers, rooms in which the hearing of speech should claim precedence over every other function. Although the acoustical design of such rooms does not ordinarily pose any difficult problems, it does require more than the specification of an absorptive material for the ceiling or for other surfaces which may be conveniently treated. For example, the control of noise is, in many locations, as important as the control of reverberation; the volume of the room should be reasonably small, and the ceiling, walls, and floor in the speaking end of the room should be designed to reflect the speakers' voices to all listening sections of the room; if the room must be a large one, an elevated seating area for the public will be helpful, and a high quality public address system may be necessary.

3. The third building we shall consider is a legitimate theater, one in which a slight excess of reverberation is a minor defect compared with the gross deficiency of sound insulation. The theater is located on a busy metropolitan street over which automobiles, trucks, buses, and trolley cars ply their noisy course, yet the auditorium of this theater is *insulated* against this terrific din, on one side wall, only by a single pair of exit doors, with quarter inch cracks around the edges of the doors. Patrons sitting anywhere on the main floor are disturbed by the noise which pours through these doors, and those who sit within 25 feet of the doors

miss whole sentences of dialogue whenever a trolley car or a noisy truck passes by. Besides these outside noises, the ventilating fans and motors add to the steady noise level in the auditorium. There is no excuse whatever to allow such obvious imperfections to destroy the utility of an auditorium. Calculations of the over-all insulation provided by any planned or completed auditorium can be made as easily as can those for reverberation, and in many instances they are fully as important.[6]

4. "Flutter" echoes in rooms are more frequently a source of disturbance or annoyance than is generally supposed. These "flutters" can be readily observed by producing impulsive sounds, as by clapping the hands, between two parallel and reflective surfaces, such as the opposite walls of a rectangular room. They are especially pronounced when one pair of opposite walls is left reflective and the other two pairs are absorptive. Under these conditions, all sounds of short duration, as percussional sounds, staccato notes, or even the consonants of speech, are repeatedly reflected as a short wave train between the parallel walls. The resulting "flutter" in large rooms, or "buzz" in small rooms, can constitute a serious acoustical defect. Such a defect was recently observed in an experimental broadcasting studio in which absorptive material was distributed over the ceiling, the side walls, and the upper half of one end wall, the other end wall being highly reflective. The flutter between the "live" end wall and the lower half of the "dead" end wall was a distinctly audible, and at times disagreeable, accompaniment to all impulsive sounds. A spotted distribution of absorptive material over the entire "dead" end wall, or irregularities in the contour of this wall, or nonparallel walls, would prevent such flutters. Further, when the "live" end wall is highly reflective, the opposite end wall should be sufficiently absorptive to compensate for lack of absorption by the "live" end, so that the decay of sound will be practically uniform in all directions in the room.

5. An example of extreme contrast between the "live" end and the "dead" end of a music room recently came to the writer's attention. The room is a large concert hall (700,000 cubic feet) with a seating capacity of two thousand. There is a large stage with concrete walls and ceiling, and as yet a rather scant supply of stage hangings. (The reverberation time for a tone of 512 cycles, measured on the rear part of the stage, with the asbestos curtain down, varied from 4 to 6 seconds, depending upon the type of stage setting used.) The forward portions of the walls and ceiling, in front of the face of the balcony and comprising about 60 percent of the total wall and ceiling area, are hardwood wainscot, hard plaster on concrete for the walls above the wainscot, and hard plaster on metal lath for the ceiling. All wall and ceiling surfaces above the balcony are treated with a highly absorptive material. The result is essentially the same as that obtained by joining a large, reverberant room to a smaller dead room; that is, the main seating area in front of, and even under, the balcony is in a very reverberant "room," and the balcony seating area is in a very dead "room." The reverberation characteristics of this large concert hall are extraordinary. In the balcony, with no audience but with heavily upholstered seats, the rate of decay of a 512-cycle tone is about 45 db a second during the first 20 db of decay, corresponding to a time of reverberation of 1.33 seconds, but the rate of decay becomes much slower during the latter stages of decay (as slow as 15 db a second). The time required for a 60-db decay of this 512-cycle tone was 3.8 seconds on the stage (with curtain up), 3.6 seconds in the middle section of the main floor, and 3.1 seconds in the balcony. The measured reverberation time is everywhere longer than that based on the rate of decay during the first 20 db of decay, and also at least a full second longer than that calculated by the usual reverberation formulas. This, of course, is the result of the non-uniform distribution of absorptive material in the auditorium. All absorptive material is confined either to the floor (covered with upholstered seats) or to the walls and ceiling above the balcony. Consequently, the reverberant sound consists first of a relatively rapid rate of decay, especially in the balcony, followed by a slower rate made up of sound traveling primarily in a horizontal direction, and confined principally in the front

[6] For details, see author's *Architectural Acoustics* (John Wiley and Sons, 1932), Chapter XIII and pp. 430–431.

portion of the auditorium, although during the latter stages of decay the reverberant space is feeding sound energy into the dead space. This defective condition was partially corrected by applying a highly absorptive material to the lower portion of the side walls and the rear wall under the balcony.

Although, as a general rule, it is advisable to design a music room such that the performers' end of the room is slightly more reverberant than the listeners' end, extremes, such as the concert hall here referred to, should be avoided.

In the preceding paragraphs we have cited several examples of faulty acoustics in rooms in which the generation or hearing of speech or music is the primary function. In all these rooms acoustical materials had been installed, and in most instances in such amounts as were indicated by generally accepted reverberation formulas and optimal times of reverberation. It should not be supposed that the examples selected are representative of results usually attained by the use of absorptive materials prescribed on the basis of the usual reverberation formulas and optimal times of reverberation. On the contrary, this simple and widely used procedure leads to satisfactory results in most instances. But if one is aware that the theory underlying such a procedure is only approximate in large rooms, and not even approximate in small rooms, and that other factors besides reverberation are pertinent to good acoustical design, one can avoid such imperfections as have been considered above, and can attain satisfactory if not ideal acoustics.

These remarks will be concluded with an appeal to do all that is reasonably possible in designing for good acoustics in buildings. The problem of acoustics begins with the selection of the site and ends with the furnishing, testing, and maintaining of the building. The necessary steps, approximately in chronological order, are as follows: (1) *Selection of a site* (theaters, schools, churches, and hospitals, especially should be located in quiet surroundings); (2) *Noise survey*, to determine the amount of insulation required to reduce the noise level in the building to a satisfactory condition; (3) *Insulation against outside noise*, which includes not only the selection of the proper sound insulative and sound absorptive constructions, but also the proper arrangement of rooms, corridors, entrances, windows, and other appurtenances of the building; (d) *Design of the shape of the room* (shapes should be designed which will not only avoid such acoustical defects as echoes,[7] interfering reflections, room flutter, and sound foci, but will facilitate the most advantageous flow of sound energy to all auditors in the room, and at the same time will preserve or even enhance the natural beauty of speech and music); (5) *Control of the noise within the building*, including solid-borne as well as air-borne noise and vibration; (6) *Selection and distribution of the absorptive and reflective materials to provide the optimal conditions for both steady state and transient sounds throughout the room*, a problem which deserves special study and careful planning, and one which involves, besides the acoustical characteristics of the materials, such properties as structural strength, decorative possibilities, adaptability to the surfaces available for, or requiring, absorptive treatment, maintenance, sanitation, ease of application, fire hazard, absorption of water, attraction for vermin, "fool-proofness," durability, and cost; (7) *Supervision of the installation of acoustical materials* (especially necessary for the application of acoustical plaster—in large buildings it is advisable to require the plastering contractor to prepare a small room for test and approval before the plaster is used in other parts of the building); (8) *Installation of amplifying equipment under the supervision of a competent engineer* is necessary in all large auditoriums—even in rooms seating as few as two or three hundred persons it will be found that many speakers with weak voices require amplification; (9) *Inspection of the finished building* should include tests to determine whether the sound insulation, the sound absorption, and the other acoustical properties have been satisfactorily attained; and (10) *Maintenance instructions, preferably in writing, should be left with the building manager*, indicating (a) how the acoustical materials can or cannot be cleaned or redecorated, (b) which furnishings in the building are essential to good acoustics, and (c) how the

[7] Echoes from the rear wall are especially common and detrimental.

humidity of large speech and music rooms should be maintained in order to avoid excessive absorption of high pitched sounds.

The foregoing steps, or their equivalent, if carefully executed, will lead to good acoustics. Developments in modern theories of room acoustics, supplemented by additional empirical data, will contribute to more reliable criteria than we now have for determining the best acoustical shape of a room and the most favorable distribution of absorptive and reflective materials throughout the room, but if proper use be made of what is now known there need be no anxiety respecting the outcome in the acoustics of buildings—the outcome will be good. How well this can be accomplished will be disclosed by other speakers who follow me in this symposium.

Planning Functionally for Good Acoustics*

J. P. MAXFIELD AND C. C. POTWIN
Electrical Research Products, Inc., New York, New York

(Received February 19, 1940)

THE question which the authors were requested to answer in this symposium was: "What constitutes good acoustics?" This question cannot be answered precisely, for the reason that subjective judgments of good acoustics vary with different individuals. The orchestra leader and musician have their conceptions of what good acoustics means, while the producer and musical critics may have different ideas. The acoustical engineer feels that his opinions should be considered and, finally, the listener must be satisfied. Good acoustics are achieved when the entertainment space is made satisfactory to both the listener and the performer. Then, and only then, can acoustics become conducive to a satisfying interchange of thought and emotion between the musician or actor and his audience.

When an auditorium or listening room is planned to have a desirable time of reverberation, a proper frequency reverberation characteristic, a careful control of the discrete first-order reflections and a background reverberation free from violently fluctuating variations in intensity level during sound die-away, it yields acoustical conditions which are praised by both the listener and the performer. Empiricism has had to play an important part in the development of methods and techniques for controlling these characteristics of sound in enclosures, for the reason that subjective judgments of good acoustics must be recognized as one of the bases for auditorium planning.

As a result of studies of the acoustical conditions surrounding the recording and reproduction of sound, several of the physical factors affecting its transmission and reception in rooms have been correlated with qualities which the artist, producer, and listener have judged to be pleasing or displeasing. These factors divide themselves into three major classifications. The first deals with the desirable amount of reverberation and its frequency characteristic for a room of given size. The second deals with the nature of the successive first-order reflections reaching the listening position, and includes the manner in which the sound in general dies away during the reverberation decay period. The third deals with certain techniques which, in the main, control the frequency characteristic of the individual first few reflections reaching the listening area and their relationship to (a) the sound coming directly from the source, and (b) the general background of reverberation.

(1) The desirable time of reverberation at 512 c.p.s. has been treated at considerable length in the literature, and there is nearly universal agreement as to the relationship of this value to the size of an auditorium and the nature of the sound to be produced. However, the situation is different regarding the desirable frequency characteristic of reverberation for auditoriums and listening rooms. MacNair derived from theoretical considerations a suggested optimum characteristic for the frequency reverberation of auditoriums.[1] Empirical practice has indicated that for large auditoriums having volumes greater than 300,000 cubic feet, the MacNair values produce exceedingly good results, particularly for music. For speech, however, somewhat lower values of reverberation time at the low frequencies may improve articulation and naturalness. A nearly flat frequency reverberation characteristic has

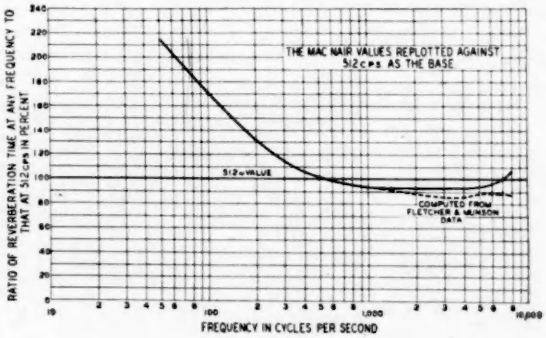
RATIO OF REVERBERATION TIME AT ANY FREQUENCY TO THAT AT 512 cps IN PERCENT
THE MAC NAIR VALUES REPLOTTED AGAINST 512 cps AS THE BASE
512 cps VALUE
COMPUTED FROM FLETCHER & MUNSON DATA
FREQUENCY IN CYCLES PER SECOND

FIG. 1.

* Presented at the Joint Symposium on Architectural Acoustics at the Meeting of the Acoustical Society of America, November 3, 1939.

[1] W. A. MacNair, "Optimum Reverberation Time for Auditoriums," J. Acous. Soc. Am. **1**, 242 (1930).

yielded the best general results for both speech and music in rooms having volumes less than 50,000 cubic feet.

Figure 1 shows the MacNair curve, replotted against 512 cycles as the base and with frequencies above 1000 c.p.s. corrected on the basis of the Fletcher and Munson data.[2] Curve *A* in Fig. 2 is based on the results of the authors' empirical studies and design practice and shows the ratio of the reverberation time at 128 c.p.s. to the time at 512 c.p.s. in percent expressed as a function of the size of the auditorium. It represents the greatest departure from the MacNair values for auditoriums of 10,000 cubic feet or less. Curve *A* in Fig. 3 is a similar expression of the values for 256 c.p.s.

A number of successful audience rooms have been planned on the basis of these data. However, the curves will probably be subject to reconsideration as the technique of controlling the individual characteristics of the reflected sound within the listening space advances. A reconsideration of the desirable times of reverberation at 512 c.p.s. may also be necessary on the same basis because experience has proved that with better control of the discrete reflections reaching the listening area, reverberation times higher than those generally recommended ten years ago can be used.

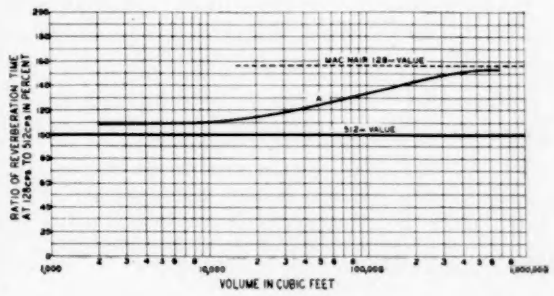

FIG. 2.

(2) The conclusion has been reached that a proper balance must be maintained between (a) the amount and character of the sound energy reaching the listening area from the first few reflections originating at the walls and ceiling, and (b) the general background of the reverberant sound. It is important that the first few reflections come in general from a large number of relatively small reflecting areas rather than from a small number of large reflecting areas. Both the reflecting and absorbing areas should vary in size and spacing and should, in general,

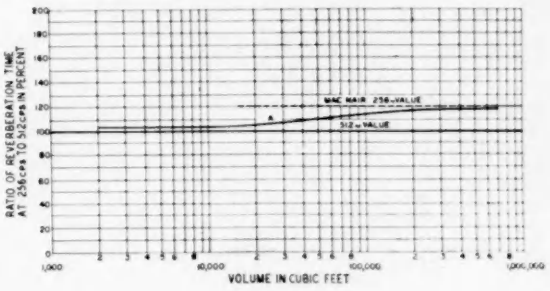

FIG. 3.

bear a nonsymmetrical relationship to each other when arranged on opposite surfaces of the auditorium.

(3) The control of the first few reflections and of the general background reverberation both by the internal shaping of surfaces and distribution of sound absorbing and reflecting areas was covered in a previous paper.[3] There are also other distinctive features of design which have proved successful in this connection and for that reason warrant consideration in this discussion. For example, in auditoriums to be used for sound motion pictures, spoken drama, or lectures, it appears desirable to keep the enclosed volume, exclusive of the stage, as low as 130 cubic feet per seat. Experience has proved that substantial savings in building costs and acoustical treatment of surfaces can be realized when consideration is given to this factor in design. Under such conditions very little, if any, sound absorbing material is needed for the walls or ceiling of the auditorium provided these surfaces are so shaped that they aid in controlling the destination and dispersion of the first few reflections.

A somewhat greater cubic foot volume per seat appears desirable in auditoriums to be used mainly for musical productions. In this instance the sensation of spaciousness resulting from the greater cubic foot volume seems to enhance the emotional effect regardless of whether music is heard directly or is reproduced in the auditorium.

[2] H. Fletcher and W. A. Munson, "Loudness, Its Definition, Measurement and Calculation," J. Acous. Soc. Am. **5**, 82 (1933).

[3] C. C. Potwin and J. P. Maxfield, "A Modern Concept of Acoustical Design," J. Acous. Soc. Am. **11**, 48 (1939).

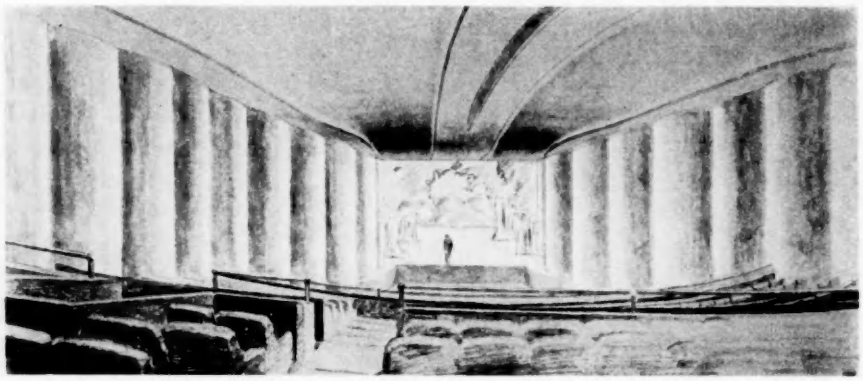

FIG. 4.

Another element of design which might fall under this classification has to do with the shaping of internal surfaces to heighten the acoustical image. Initially, a part of the wall or ceiling area of the auditorium is shaped to reflect sound diffusely to the audience area. These surface forms are coordinated with the nonsymmetrical arrangement of sound absorbing material so that their combined action also helps to form and maintain the "around-the-room" reflections desirable for background reverberation. The remainder of the interior surfaces of the auditorium are then shaped to direct high frequency reflections to audience areas where the intelligibility of speech and the definition of music are most likely to become submerged in the background reverberation.

Such a design produces what might be termed a "diffuse whispering gallery" effect. In other words, taking the auditorium as a whole, the sound energy is in a generally diffuse state, yet portions of the surfaces are projecting the higher frequencies to the seats farthest removed from the stage.

Perhaps one of the best illustrative examples of a building in which these various principles of design and acoustical treatment have been employed is the new Theater and Arts Addition to the Memorial Union recently completed at the University of Wisconsin. Through the foresight of the architects, their theater consultant and the University, acoustics was placed on an equal footing with the other factors of basic planning for the purpose of fulfilling functional, aesthetic and social needs. Acoustically, the problem presented was one of designing a building adaptable to the following functions.

(a) To provide entertainment to as many as 1300 persons in the form of spoken drama, sound motion pictures, and symphony music, within the confines of a single space.

(b) To provide facilities for study and instruction in drama and radio broadcasting technique, and for general lectures, small recitals, and sound pictures within a space having a seating capacity not to exceed 200, one of the special requirements being that the stage might be opened up to partially surround the seating area.

(c) To provide further facilities for radio broadcasting and for the study of small studio and phonograph techniques, some of these spaces to be used as multipurpose rooms.

(d) To provide facilities for two workshops, one to be used for the design and construction of stage scenery, and the other for the study of crafts.

(e) To provide for a bowling area in the basement of the building, this area to contain a total of eight alleys.

It was further stipulated that any or all of these facilities should be usable simultaneously without noise interference from one to the other.

The plan was developed with (1) a theater seating approximately 1300 persons which might under certain conditions of use be reduced in seating capacity to 700 persons; (2) a space called the Play Circle seating about 200 persons, wherein the speech and action of the drama could be studied on a stage partially surrounding the seating area, this audience room to be usable also as a broadcasting theater; (3) a small broadcasting studio and phonograph room, both adjacent to the Play Circle and so designed that they could also be used as viewing rooms for the

Play Circle; (4) a workshop immediately behind the stage of the theater, to be used for the construction of sets, scenery and other stage accessories; (5) similar workshops for study of the crafts on the floor above the Play Circle, and (6) the bowling alleys in the basement one level below the theater.

Basically, the design of the building presented a special problem in sound isolation since it involved the segregation of high noise levels from necessarily quiet areas within the limits of a very compact plan. The floor of the bowling space was completely isolated acoustically from the main building structure. This isolation was accomplished economically through the use of cork and the more common flooring materials. It prevents the transmission of structure-borne sound and vibration to other parts of the building, a difficulty commonly experienced in cases where suitable isolation is not provided at the source. The transmission of air-borne noise in the higher frequency range is minimized through a careful arrangement of dividing walls and ceilings.

One special requirement was that no sound absorbing materials should be used in the walls or ceiling of the alley space. For that reason the average noise level measured in the space was 95 db, with peak levels reaching 115 db during pin strike. Yet both sound level measurements and hearing tests indicate that this noise is not transmitted into any of the spaces where quiet conditions are essential. Through the use of double wall constructions and floors of the same type as those employed for the alleys, the high noise levels generated by band saws and other machinery in the workshops have also been confined to the spaces wherein they originate.

The Theater and the Play Circle offered unusual and interesting problems in the shaping of interior surfaces and the placement of absorbing materials. The basic form of the theater interior was developed with a balcony in order to best fulfill the visual and acoustical needs. From the proscenium opening, the walls extend toward the rear with a series of convexly curved forms for sound diffusion, as shown in the panorama view in Fig. 4. These forms taper off gradually to flat surfaces at points where the principal areas of acoustical treatment begin. The ceiling, as shown on the section in Fig. 5, follows a form of gradually decreasing curvature from the stage opening to the rear of the auditorium and meets the side walls at right angles. This ceiling tends to carry the high frequencies, on their first reflection, toward the rear of the balcony, thereby enhancing the tone color of music and the intelligibility of speech in this area.

The wide range of uses for which the theater was intended made it advisable to allow 170 cubic feet per seat in the basic design of the auditorium, resulting in a total volume of 225,000 cubic feet. In spite of this comparatively large volume per seat, only 750 sound absorption units of acoustical material were used for the walls to make the auditorium suitable for either direct

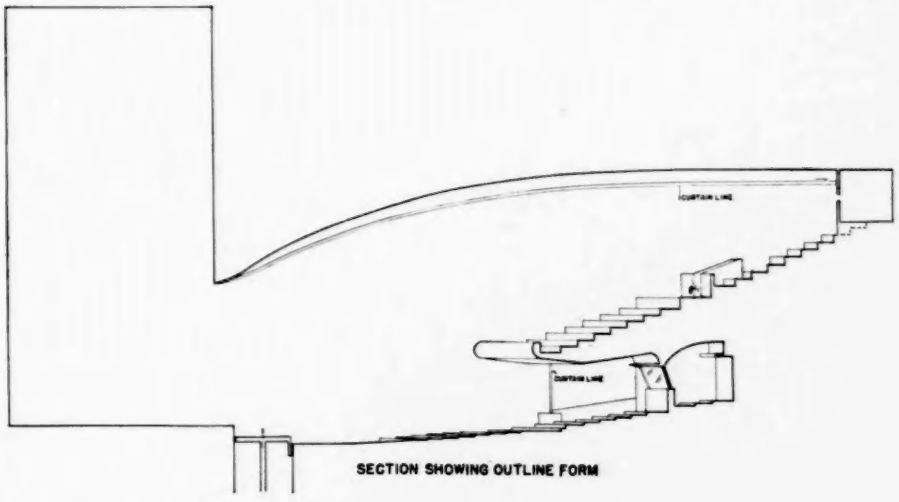

FIG. 5. Wisconsin Union Theater, University of Wisconsin.

or reproduced speech and music. The acoustical material is distributed in panels over the wall surfaces. The panels along the side walls are so arranged that they bear a nonsymmetrical relationship to each other on opposite walls and coordinate with the action of the diffusing surfaces to aid in controlling the general background reverberation. Furthermore, the panels are recessed so that their non-uniform arrangement works out well with the decorative treatment of the interior.

The authors believe, in the light of present knowledge, that an auditorium of this size, which must necessarily be designed for widely differing acoustical functions, requires a small amount of sound absorbing material installed on the wall or ceiling surfaces. It may very well be that as the control technique advances, auditoriums of this size or greater may be designed with no sound absorbents needed other than the seats and the audience.

There are two other unusual features which should prove interesting in connection with the design of the theater. The first is a sliding partition of double plate glass at the rear of the mezzanine section on the main floor which acoustically isolates a part of the seating area from the main auditorium, thus providing facilities for the study and discussion of stage productions while they are in progress. The second is the use of especially selected dividing curtains which may be drawn across the complete mezzanine section and the rear center section of the balcony to reduce the apparent size of the auditorium for small audiences without affecting hearing conditions materially within the remaining sections.

The percentage syllable articulation[4] as determined at the completion of the theater, using a caller positioned on the stage and ten observers seated in five different positions throughout the audience area, was 89 percent. The value determined for the extreme rear of the balcony section alone was 84 percent.

The Play Circle, sketches of which are shown in Fig. 6, is unique in its design because of the stage which extends around to both sides of the seating area. Through the use of sliding doors, the side sections of the stage can be closed off, thereby forming the conventional proscenium opening. When the architectural and acoustical requirements are fulfilled in a design of this type, it proves exceedingly flexible from the operating viewpoint and enables the users to obtain a wide variety of dramatic effects within a relatively small audience room. The semi-circular type of stage with sliding doors for unit division also provides facilities for striking a set on one side while the other side is in use.

The desire to create a visual sense of space through the use of special methods of lighting necessitated the design of a stage rear wall in

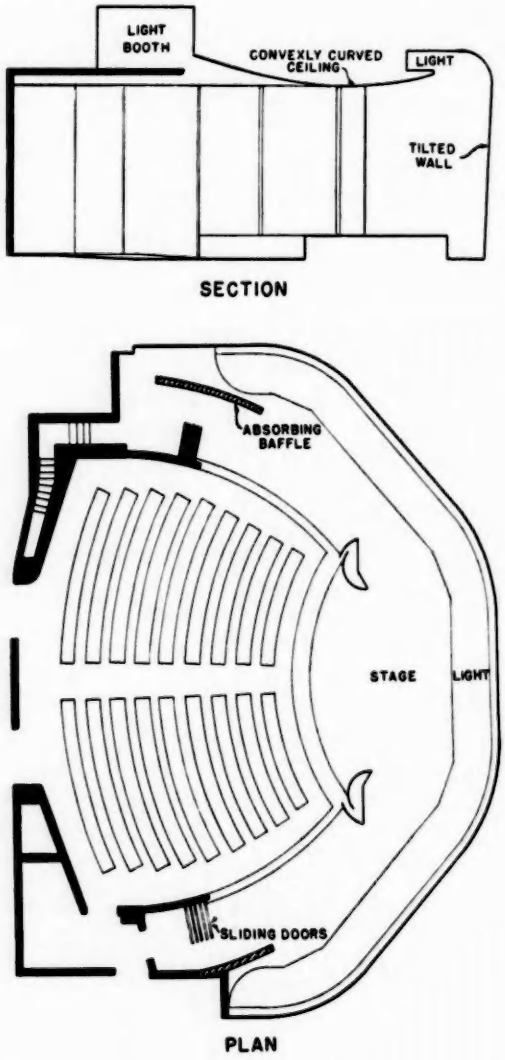

FIG. 6.

[4] H. Fletcher and J. C. Steinberg, "Articulation Testing Methods," Bell Sys. Tech. J. (October, 1929); Also J. Acous. Soc. Am. (Supplement to January, 1930).

the form of a cyclorama which would give the effect of following the curved contour of the stage floor. In order that the acoustical requirements might also be satisfied, the wall was designed in three straight segments with connecting curvatures of very short radius between. This design greatly reduced the high levels of sound concentration which would have resulted had a completely curved wall of concave form been provided at this point. Secondly, the entire cyclorama was tilted back from the vertical so that discrete reflections and short foci of sound would be thrown into the unoccupied space above the heads of the auditors.

Sound-absorbing baffles are provided at each side of the stage as shown in Fig. 6 to further aid in controlling sound reflections and to reduce conversational noise backstage during the time performances are in progress. The rear wall behind the seating area is also treated acoustically with a limited amount of sound absorbing material well distributed over the surface. Tests made in this room after its completion indicated a percentage syllable articulation of 87 percent.

In view of the consideration now being given to the distribution of sound in enclosed spaces, it would appear that a complete solution of the problem will soon be forthcoming. The work of Knudsen, Meyer, Hunt, Morse, Bolt and others[5–13] is already pointing the way to definite methods of analysis which should lead to the removal of empiricism from the physical part of the present technique. Such a solution of the problem would establish a firm basis for the advances which have already resulted from the empirical technique and would make it possible definitely to limit the use of this technique to the correlation of the physical factors with the subjective ones.

[5] V. O. Knudsen, "Resonance in Small Rooms," J. Acous. Soc. Am. **4**, 20 (1932).

[6] V. O. Knudsen, "Recent Progress in Acoustics," J. Soc. Mot. Pict. Eng. **29**, 233 (1937).

[7] E. Meyer, "Reverberation and Absorption of Sound," J. Acous. Soc. Am. **8**, 155 (1937).

[8] F. V. Hunt, "Investigation of Room Acoustics by Steady-State Transmission Measurements," J. Acous. Soc. Am. **10**, 216 (1939).

[9] F. V. Hunt, L. L. Beranek and D. Y. Maa, "Analysis of Sound Decay in Rectangular Rooms," J. Acous. Soc. Am. **11**, 80 (1939).

[10] P. M. Morse, "Some Aspects of the Theory of Room Acoustics," J. Acous. Soc. Am. **11**, 56 (1939).

[11] R. H. Bolt, "Frequency Distribution of Eigentones in a Three-Dimensional Continuum," J. Acous. Soc. Am. **10**, 228 (1939).

[12] R. H. Bolt, "Normal Modes of Vibration in Room Acoustics: Angular Distribution Theory," J. Acous. Soc. Am. **11**, 74 (1939).

[13] R. H. Bolt, "Normal Modes of Vibration in Room Acoustics: Experimental Investigations in Nonrectangular Enclosures," J. Acous. Soc. Am. **11**, 184 (1939).

Sound Diffraction and Absorption by a Strip of Absorbing Material

JOHN R. PELLAM
Eastman Laboratory, Massachusetts Institute of Technology, Cambridge, Massachusetts

(Received February 10, 1940)

An exact solution for the sound energy absorbed by an absorbing panel set in the surface of a large, rigid, plane wall is obtained. The ability of the strip for absorbing sound is given by a "shape factor" A, defined as the ratio of the energy retained by the panel to the energy which would be absorbed, at normal incidence, by the same section of an infinite wall, covered uniformly with the same material. Curves of A are given for three angles of incidence, and for a wide range of wave-lengths and of normal acoustic impedance of the absorbing material. The "shape factor" reveals that a given material composing a panel absorbs most efficiently when the panel-width approximates, or slightly exceeds, a half-wave-length of the incident sound. For such widths, the factor is larger than two in the case of very "soft" materials, and reduces to unity for less and less absorbent panels.

INTRODUCTION

RECENT investigations in the theory of room acoustics indicate that sound waves are affected by the presence of absorbing material; and, conversely, that the absorbing properties of a material are affected by the shape and size of the containing room, and by the shape and distribution of the absorbing panels. The effect of room shape on sound decay has been begun.[1] This paper is a preliminary study of the effect of the size of the absorbing panel on the effective absorption coefficient of the panel material. The simplest case is studied, that of a uniform strip of absorbing material placed in an otherwise totally reflecting, plane wall of a room. The diffraction effects of the strip are computed, and the effect of the diffraction on the absorbing properties of the strip obtained.

More specifically, this investigation involves sound absorption and scattering by an infinitely long strip, of width d and normal acoustic impedance z, set in the surface of an infinite, rigid, plane wall.

Using Mathieu functions, P. M. Morse and P. J. Rubenstein have given[2] recently a complete solution of diffraction by rigid ribbons and by slits, and, similarly, these newly computed functions are employed in the present analysis, where nonrigid surfaces are substituted for the open slit.

SOLUTION OF WAVE EQUATION

The coordinate system, shown in Fig. 1, is given by the transformations

$$x=(d/2)\cosh(\epsilon)\cos(\phi),$$
$$y=(d/2)\sinh(\epsilon)\sin(\phi). \tag{1}$$

Solutions of the wave equation in these coordinates are even and odd Mathieu functions of angular and radial types, Se_m, So_m and Re_m, Ro_m, respectively. The z coordinate disappears, since only sound striking the wall with propagation vector perpendicular to the axis of the ribbon (i.e., in the x–y plane), is considered.

The orthogonal angular functions, $Se_m(c, \cos\phi)$ and $So_m(c, \cos\phi)$, are normalized by

$$\int_0^{2\pi}[Se_m]^2 d\phi = N_m, \quad \int_0^{2\pi}[So_m]^2 d\phi = N_m', \tag{2}$$

and the even radial functions approach the following limiting values as $\epsilon \to 0$,

$$Re_m^{I}(c,1)=\frac{1}{(2\pi)^{\frac{1}{2}}\lambda_m}, \quad Re_m^{II}(c,1)=-(2\pi)^{\frac{1}{2}}\mu_m,$$
$$[(d/d\epsilon)Re_m^{I}(c,\cosh\epsilon)]_{\epsilon=0}=0, \tag{3}$$
$$[(d/d\epsilon)Re_m^{II}(c,\cosh\epsilon)]_{\epsilon=0}=(2\pi)^{\frac{1}{2}}\lambda_m,$$

where (λ_m) and (μ_m) are constants defined in

[1] P. E. Sabine, J. Acous. Soc. Am. **10**, 1 (1938); P. M. Morse, J. Acous. Soc. Am. **11**, 56 (1939); N. B. Bhatt, J. Acous. Soc. Am. **11**, 67 (1939); R. H. Bolt, J. Acous. Soc. Am. **11**, 56 (1939).

[2] P. M. Morse and P. J. Rubenstein, Phys. Rev. **54**, 895 (1938).

reference 2, and where the substitution, $c=\pi d/\lambda$, has been used.

A simple harmonic pressure wave, traveling with propagation vector in the x–y plane and making an angle u with a totally reflecting plane wall, lying in the x–z plane, can be expressed by[2]

$$p_f=(8\pi)^{\frac{1}{2}}p_{0f}e^{-2\pi\nu it}\sum_m i^m(2/N_m)Se_m(c,\cos\mu)Se_m(c,\cos\phi)Re_m^{I}(c,\cosh\epsilon) \quad (4)$$

(remembering the properties of even and odd functions).

If an absorbing strip of normal acoustic impedance z, assumed real, and width d, is set in the surface of the wall (its axis coincident with the z axis, and its edges determining the foci of the confocal, elliptical cylinders described by the coordinate system (1)), a scattered wave is set up to satisfy the boundary condition at the absorbing surface. This condition is

$$v_n=\frac{1}{2\pi\nu i\rho}\frac{\partial p}{\partial n}$$

or

$$\frac{\partial p}{\partial\epsilon}=i\left(\frac{\pi d}{\lambda}\right)\left(\frac{\rho c}{z}\right)|\sin\phi|\,p, \quad (5)$$

where (ρc) is the specific acoustic impedance of air, λ is the wave-length of the sound, n the normal to the strip, and v_n the normal particle velocity.

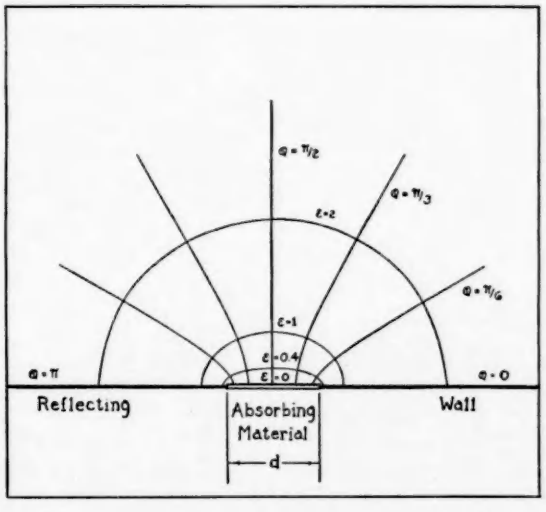

FIG. 1. Orthogonal coordinate system employed; composed of elliptic and hyperbolic cylinders, shown in cross section.

The resulting pressure wave becomes

$$p=(8\pi)^{\frac{1}{2}}p_{0f}e^{-2\pi\nu it}\sum_m i^m(2/N_m)Se_m(c,\cos\mu)Se_m(c,\cos\phi)[Re_m^{I}(c,\cosh\epsilon)+A_mRe_m^{III}(c,\cosh\epsilon)], \quad (6)$$

where the complex function

$$Re_m^{III}(c,\cosh\epsilon)=Re_m^{I}(c,\cosh\epsilon)+iRe_m^{II}(c,\cosh\epsilon)$$

provides for outgoing waves, and the coefficients A_m, essentially real quantities, are determined by the following relations:

$$A_k=\sum_m\alpha_{mk}[1+A_m(1-i2\pi\mu_m\lambda_m)], \quad (7)$$

$$\alpha_{mk}=\frac{i^{m-k}}{N_m}\left(\frac{\pi d}{\lambda}\right)\left(\frac{\rho c}{z}\right)\frac{Se_m(c,\cos\mu)}{Se_k(c,\cos\mu)}2\pi\lambda_m\lambda_k\times\int_0^{2\pi}|\sin\phi|\,Se_m(c,\cos\phi)Se_k(c,\cos\phi)d\phi.$$

Since the calculations of the A's are tedious, we have made them only for real values of the impedance z.

Absorption of Energy

The average rate of energy absorption $\bar{E}$ is given by the integral, over the strip surface, of $p\bar{p}/2z$, where $\bar{p}$ denotes the complex conjugate of p, and where the normal acoustic impedance z, is real and, for a given frequency, independent of the wave form.[3] Hence

$$\bar{E}=\frac{p_{0f}^2}{(\pi d/\lambda)}\frac{4\pi d}{(\rho c)}\sum_m[A_m+A_m\bar{A}_m]\frac{Se_m^2(c,\cos\mu)}{N_m}, \quad (8)$$

A_m being effectively real for small absorption, i.e., if the normal absorption coefficient for the material does not exceed about 0.5.

Results

Values of the definite integral

$$I_{mk}=\int_0^{2\pi}|\sin\phi|\,Se_m(c,\cos\phi)Se_k(c,\cos\phi)d\phi$$

and of the α's were computed and the coefficients A_m then determined, by a method of successive

[3] F. J. Willig, J. Acous. Soc. Am. **10**, 257(A) (1938); F. V. Hunt, J. Acous. Soc. Am. **10**, 216 (1938); P. M. Morse, J. Acous. Soc. Am. **11**, 56 (1939).

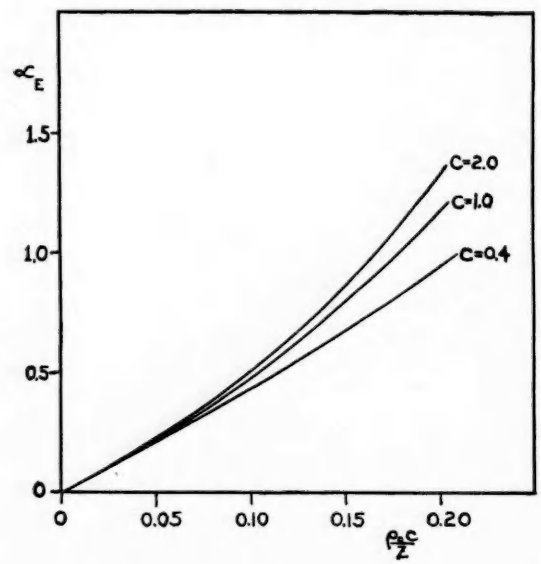

FIG. 2. Typical curves of effective absorption coefficient (α_E) as a function of $\rho c/z$. (Normal incidence, $u=90°$.)

approximations, for three values of the angle u (0°, 60°, and 90°) and for a wide range of real values of $z/\rho c$. From these tables the absorbed and scattered energies for these cases could be computed.

The first quantity computed is the "effective absorption coefficient" of the strip material, given by the ratio of the energy absorbed by the strip to the energy transmitted by a width d of incident wave. In other words, it is the ratio of the absorbed energy to the energy which would strike the strip at normal incidence on the basis of geometrical optics. The formula for this quantity is:

$$\alpha_E=\frac{8\pi}{(\pi d/\lambda)}\sum_m [A_m+A_m{}^2]\frac{Se_m{}^2(c,\cos\mu)}{N_m}. \quad (9)$$

Typical curves are given in Fig. 2 and show that, for small absorption, the energy retained by the strip is nearly proportional to $\rho c/z$.

A more useful quantity, however, is the "shape factor" A, the ratio of the amount of energy actually absorbed by the strip to the energy the same width of material would absorb if it were part of an infinite sheet of absorbing material. Morse[4] has shown that the absorption coefficient for normal incidence of an infinite plane wave on an infinite plane wall, of material with normal impedance z, is

$$\alpha_f=\frac{4\rho c/z}{[1+(\rho c/z)]^2}. \quad (10)$$

The ratio of the absorption coefficient α_E for the strip to α_f is the "shape factor" A.

$$A=\left(\frac{\alpha_E z}{4\rho c}\right)\left[1+\left(\frac{\rho c}{z}\right)\right]^2. \quad (11)$$

It is the ratio of energy actually absorbed to the energy which geometrical optics predicts would would be absorbed by the strip at normal incidence. Curves of A are given in Figs. 3, 4, and 5, plotted against (d/λ) for several values of $\rho c/z$.

In practice, the normal acoustic impedance z increases with frequency for most materials, especially those which are highly absorbent. Therefore, the curves of A, as shown, should be taken to describe the variations in the "shape factor" when the panel-width d is varied, keeping the wave-length constant at any convenient value.

CONCLUSIONS

The curves for the "shape factor" A show that a strip of material which is poorly absorbent is hardly more effective in narrow strips than it is in a single large sheet. On the other hand, material having "absorption coefficient" greater than 0.5 is more than twice as effective when cut in strips a half-wave-length wide than it is in large sheets. This result seems reasonable, for the more absorbent the material is, the more it will distort the wave and the more energy it will "pull" in from its neighborhood.

Presumably this trend would reverse itself for very large values of $(\rho c/z)$, particularly if the z has a reactive component; for theory shows that as $(\rho c/z)$ is increased, the wave is first "attracted" to the absorbing surface, and then, for very small z, is repelled. This reversal, which would come only for very "soft" materials, is indicated in the few calculations made using such values of $\rho c/z$; since the accompanying curves, however, cover only the majority of impedances encountered in practice, they do not show this effect.

[4] P. M. Morse, *Vibration and Sound* (McGraw-Hill, 1936), pp. 303–304.

The "shape factor" is different for different angles of incidence. A "resonance" peak occurs for normal incidence when the strip is half of a wave-length wide. For wider strips the factor decreases, eventually reaching unity for strips of infinite width. At grazing incidence the resonance reinforcement of the diffracted wave is altered. There are slight depressions in the curves for A at $d=\lambda/2$ (probably corresponding to a tendency for destructive interference of the waves scattered from the two edges in this case). The highest point in the curve comes for strips somewhat wider than a wave-length. The curves for intermediate angles of incidence are intermediate in shape.

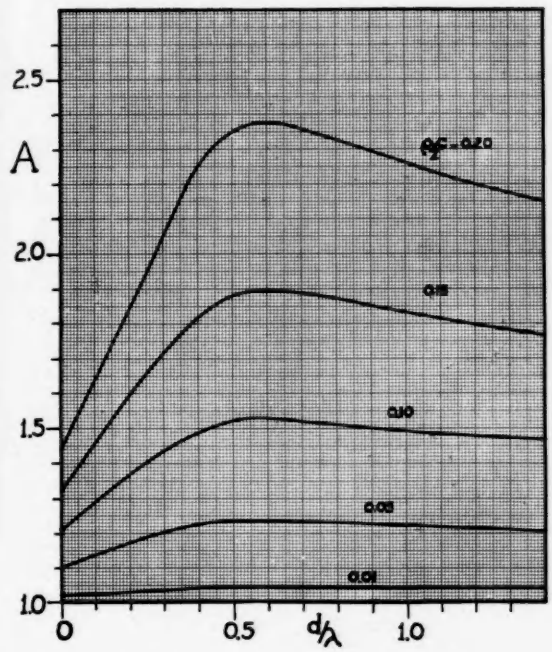

FIG. 3. The "shape factor" (A) as a function of ratio of panel-width (d) to wave-length (λ), for normal incidence ($u=90°$) (and for several values of $\rho c/z$).

It is of interest to compare these curves with the ones given[2] by Morse and Rubenstein for an open slit (their Fig. 4). In this case the factor T, which corresponds to A, goes to infinity as (d/λ) goes to zero. This is due to the fact that a very narrow slit allows a proportionately large amount of air to flow through it. A narrow strip of absorbing material, however, cannot allow such quantities of air to flow, so that A decreases again as (d/λ) becomes smaller than $(\frac{1}{2})$. We can therefore predict that if a wall is to be made absorbent by punching holes in it, a very large number of exceedingly small holes will be much more absorbent than a single large hole. On the other hand, if the absorption is to be provided

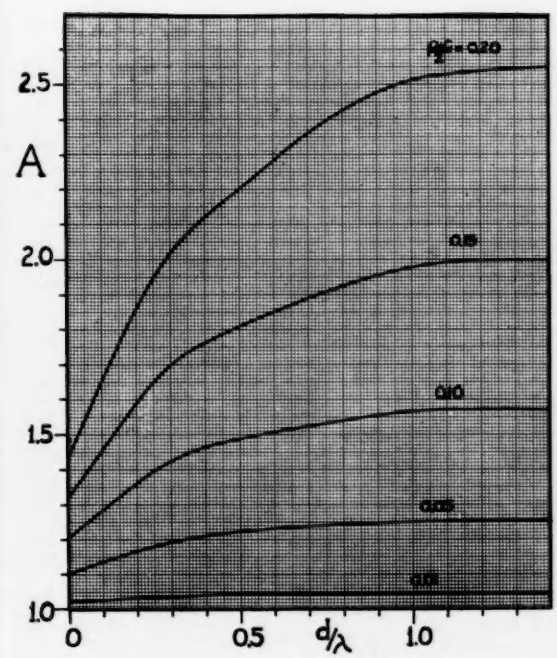

FIG. 4. "Shape factor" as a function of (d/λ), for oblique incidence ($u=60°$).

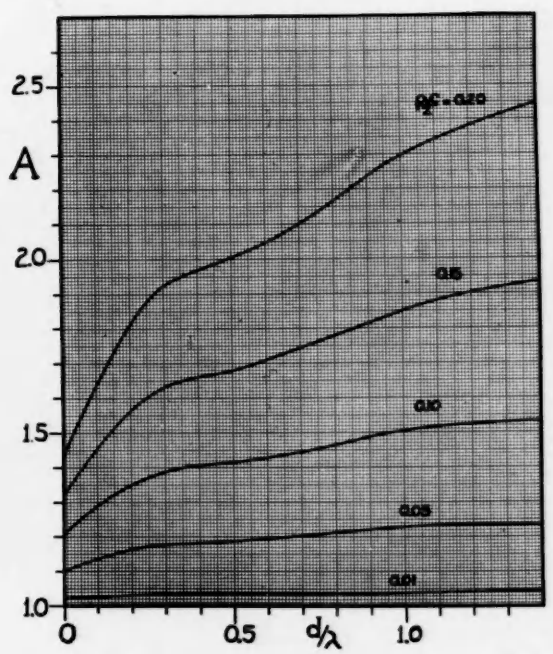

FIG. 5. "Shape factor" (A) as function of (d/λ), for grazing incidence ($u=0°$).

by pieces of absorbing material, the most effective procedure is to place on the wall, in an irregular pattern, pieces of material whose dimensions are between half of a wave-length and a wave-length, rather than to concentrate all the absorbing material in one large panel.

The results computed here are for plane waves striking an absorbing strip in an infinite plane wall, and will be modified somewhat when the strip is on one wall in a room. However, as long as the strip is not too close to a corner (about five wave-lengths away or farther), and as long as the strip is five or more wave-lengths long, the curves for A should not be greatly changed.

The author wishes to express gratitude to Professor P. M. Morse not only for his suggestion of this problem, but also for his guidance and help in its solution.

Measurement of Impact Sound Transmission Through Floors

ROBERT LINDAHL AND HALE J. SABINE
The Celotex Corporation, Chicago, Illinois
(Received January 22, 1940)

THE subject of impact noise transmission is one of considerable practical importance.

We know this because of the problems which constantly arise as of a tap dancing school over a restaurant, a gymnasium over a class room, a bowling alley over a theatre, a warehouse over an office and young children in apartments over elderly people. There is a crying need for practical test data, especially to solve the problem in low cost residence and apartment construction.

Just how to measure the impact transmitting properties of a floor, and how to express the results in figures by which various constructions can be rated and compared intelligently, is admittedly a difficult problem. As far as we know, the National Bureau of Standards should be given the credit for being the first and only American testing laboratory which has published the results of measurements of impact noise. From a practical standpoint, however, although a large number of floor constructions have been tested, there is still need of additional data to solve the every day problems which arise. In the National Bureau of Standards method, a mechanical tapping machine is placed on the floor panel under test, and the noise level set up by the impacts is measured both in the room above and in the room below the floor. The difference between the two noise levels is taken as a measure of the impact sound insulating efficiency of the floor construction, expressed in one single figure.

There are frequently situations which arise for which such a single rating may not be entirely adequate. Our experience indicates that one must consider separately the noise transmitted through the floor and the noise created above the floor. By using a single figure giving only the difference in noise level above and below the floor, it is impossible to compare the noise transmitting properties of two constructions independently of their noise emitting characteristics in the room above.

It is often desirable in solving actual problems to compare not only the total noise transmitted by different floor constructions, but also the frequency characteristics of the noise. Impact noise heard through a floor is by its very nature particularly annoying, and the annoyance factor very probably depends quite as much on the quality of the noise as on its over-all loudness level. A shoe dropped by your upstairs neighbor at 2 A.M. on the carpet does not arouse as violent homicidal impulses as does the other shoe dropped on the bare wooden floor, even though the total loudness of the two noises may be the same. The greater annoyance in the latter case is due to the sharp, distinct, "near-by" quality of the noise as contrasted with the dull, muffled, remote quality of the impact on the carpet. This muffling effect depends on the attenuation of the high frequency components of the impact noise, and it is for this reason that the transmission characteristics at various frequencies must be considered. To the best of our knowledge, frequency analyses have not previously been applied to the case of impact transmission.

In the following, we will describe a method of measurement which we have used to arrive at more complete and definite information on floor constructions. First, however, it would be well to analyze the problem. What one would like to obtain is a single figure for each floor construction that represents the characteristic impact transmitting properties of that construction, and involves no other variable, just as the transmission loss of a wall denotes its characteristic insulating efficiency against air-borne sound. This figure should be arrived at in such a way that it can be duplicated with reasonable accuracy by different laboratories or in field tests. Finally, what is wanted is a figure which can be correlated with the relative performance of different constructions under actual conditions of use.

One might use as the basis for such a figure the absolute decibel level of the noise transmitted when the floor is subjected to impact. This noise level would depend first, of course, on the floor

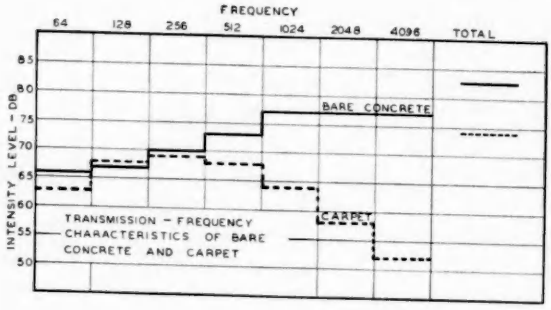

FIG. 1.

construction itself, including the surfacing material. It would also depend on the force of the impact, the hardness of the object striking the floor, the acoustical properties of the room below the floor, and on the position of the pick-up microphone. In the case where one measures the average noise level set up by a regular succession of impacts, the readings would depend on the frequency of the impacts and on the damping characteristics of the meter movement. There are other possible variables, such as the application of the impact through a static load, as when vibration is conducted through the feet of a piano or radio cabinet, and the area of panel under test.

Every one of these variables would have to be arbitrarily standardized and accurately controlled in order for the same floor construction to give the same noise level in different laboratories or test set-ups. This standardization is theoretically possible, but would be difficult to achieve practically. An alternative is to use a comparison method, by which the noise transmitted by each floor construction is compared to that transmitted by an arbitrarily selected standard floor. In this way, all of the above-mentioned variables can be canceled out, providing the standard floor is tested along with the other constructions, and under exactly the same conditions. This is the method we have employed in a recent series of tests, and which is described as follows:

All of the tests were made on an existing 6-inch concrete slab floor separating two stories of a warehouse. The concrete slab was exposed in the room below. The tests were made on a number of small panels 24 to 27 inches square of various flooring materials and combinations of materials, all applied to the bare concrete floor and adjacent to each other. The panels were all laid in the same large room, and the total area covered was above a single large room on the floor below.

The device for producing the impacts was a portable tapping machine similar to that used by the National Bureau of Standards. Steel hammers weighing about two pounds were operated by a motor driven cam shaft so as to deliver 440 impacts per minute, or a little over 7 per second. The tapping machine was moved from one panel to the next, the noise level in each case being measured in the room below.

The noise measuring apparatus consisted of a General Radio sound level meter and a set of

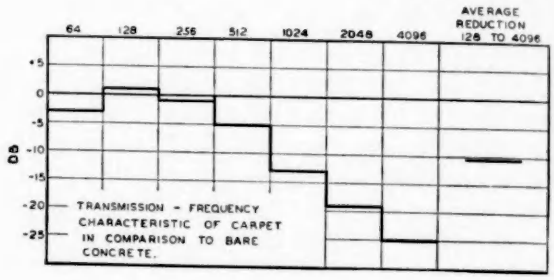

FIG. 2.

General Radio band-pass filters, which were inserted in the meter circuit. The filters were designed to pass any one of 7 octave-wide frequency bands having their centers at the 7 frequencies 64, 128, 256, 512, 1024, 2048 and 4096 cycles, thus affording a frequency analysis in octave steps of the entire noise spectrum. For each floor panel tested, readings of the over-all noise level and of the noise level in each of the 7 frequency bands were made in the room below, with the microphone placed 3 feet from the ceiling directly below the point of impact. The impacts were frequent enough, and the room below was sufficiently reverberant to insure reasonably steady readings of the meter. All readings were taken with the sound level meter set on the "*C*" or flat frequency weighting network, so that the readings were recorded as intensity levels.

In addition to the measurements on the various panels, readings were taken for impacts made directly on the bare concrete floor surface. This was arbitrarily taken as the standard floor, and the values obtained for each of the flooring panels, at each frequency band, were stated as a difference in decibels compared to the levels

measured for the bare concrete. Previous tests had shown a bare concrete floor to be as good a practical standard as could be obtained. Fig. 1 shows the measured frequency analyses and the over-all levels of the noise transmitted by the bare concrete and by the concrete covered with a panel of heavy unlined carpet. This shows that at least for the particular type of impact used in these tests, the concrete transmitted noise having most of its energy in the high frequencies. This noise as heard in the room below was very sharp and distinct, and exceedingly annoying. With the carpet the high frequencies were greatly attenuated, and the noise sounded dull, muffled, more remote and much less annoying. Fig. 2 shows the comparative transmission characteristics of the carpet with the noise levels for the bare concrete taken as zero at each frequency.

As the tests were made, the observer in the room below attempted to compare subjectively the annoyance or "noisiness" of the noise transmitted by the various panels, and to record his impressions. In comparing these notations with the test data, it was found that in general the comparative "quietness" of the panels depended more on the reduction of the high frequency components than on the reduction of the total noise. In order to give proper credit to the attenuation of the high frequencies, it was decided to use as a rating for each panel an average decibel reduction arrived at by taking

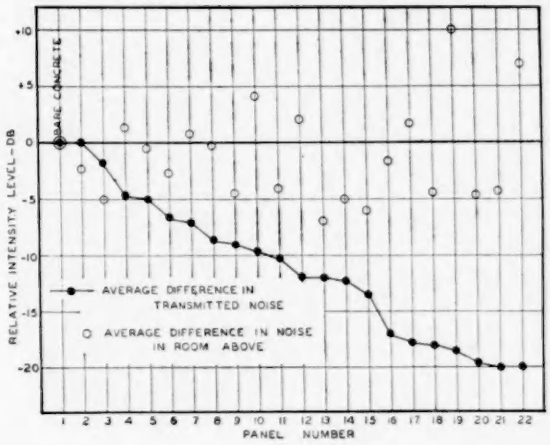

FIG. 3.

the numerical average of the reductions in the 6 frequency bands centering at 128 to 4096 cycles. The reduction at 64 cycles was omitted from the average because it did not vary appreciablywith different constructions, and did not appear to affect the relative quietness as judged by the ear. This is the same method as is used in obtaining the average transmission loss of a floor or partition for air-borne sounds.

Table I shows the reductions in transmitted noise for a number of the typical constructions in the panels tested. The comments of the observer below are shown in the tabulation to indicate the general relationship between his impressions and the actual numerical average

TABLE I. *Noise transmitted by floor constructions compared to bare concrete.*

Order of Noisiness	Floor	Comment of Observer	Sound Intensity Level, as Compared to Bare Concrete, in Decibels						
			Frequency						
			128	256	512	1024	2048	4096	Average
1	Bare concrete	Bad	0	0	0	0	0	0	0
2	$\frac{5}{32}$" asphalt tile	Nearly as bad	+1	+7	− 5	+ 2	− 2	−13	0
3	$\frac{1}{8}$" asphalt saturated felt	Nearly as bad	−2	+2	+ 1	− 2	− 5	− 5	− 1.8
4	$\frac{3}{16}$" rubber tile	Better than concrete	+3	+1	− 2	− 5	−11	−22	− 6.7
5	Heavy carpet (no pad)	Good	+1	−1	− 5	−13	−19	−25	−10.3
6	$\frac{3}{16}$" linoleum on felt	As good as carpet	−1	−2	− 5	−12	−25	−27	−12
7	$\frac{1}{2}$" asphalt saturated celotex		+1	+2	− 7	−19	−27	−24	−12.3
8	$\frac{3}{16}$" hard board over $\frac{1}{4}$" celotex	Better than carpet	+3	−3	−19	−27	−27	−29	−17
9	$\frac{3}{4}$" wood floor on 2×3 sleepers	Very quiet, but distinct tap heard	0	−4	−21	−26	−28	−32	−18.5
10	$\frac{1}{2}$" cork tile	Very good—very low pitched sound	+2	−1	−19	−34	−33	−30	−19.5
11	$\frac{3}{4}$" wood floor on 2×3 sleepers, rock wool fill	About the same as without rock wool	0	−6	−23	−27	−32	−32	−20

TABLE II. *Noise created in room above by floor constructions compared with bare concrete.*

ORDER OF NOISINESS	FLOOR	SOUND INTENSITY LEVEL, AS COMPARED TO BARE CONCRETE, IN DECIBELS						
		FREQUENCY						
		128	256	512	1024	2048	4096	AVERAGE
1	$\frac{3}{4}$″ wood floor on 2×3 sleepers	+8	+27	+14	+10	+3	−1	+10.1
2	Same, with rock wool fill	+10	+15	+11	+6	+3	−2	+7.1
3	Bare concrete	0	0	0	0	0	0	0
4	$\frac{3}{16}$″ hard board on $\frac{1}{4}$″ Celotex	−3	+1	−1	0	−2	−5	−1.7
5	$\frac{5}{32}$″ asphalt tile	−1	+3	0	−2	−6	−8	−2.3
6	$\frac{3}{16}$″ rubber tile	−1	+2	−1	−3	−5	−8	−2.7
7	Heavy carpet (no pad)	−2	−1	−3	−7	−7	−5	−4.1
8	$\frac{1}{2}$″ cork tile	−3	0	−7	−4	−6	−8	−4.7
9	$\frac{1}{8}$″ asphalt saturated felt	−5	−2	−3	−4	−8	−9	−5.1
10	$\frac{1}{2}$″ asphalt saturated Celotex	−4	0	−4	−5	−8	−10	−5.1
11	$\frac{3}{16}$″ linoleum on felt	−3	−3	−6	−9	−9	−12	−7

taken as a measurement of the relative efficiency.

It is to be noted that a hard-surfaced construction, consisting of a composite floor, such as hardboard on fiberboard, or even a floating wood floor on sleepers, is more efficient than a resilient carpet or linoleum surface which are often considered to be the most efficient in cutting out the sound transmitted through a floor by impact.

Table II shows the results of the tests repeated in the room above with the microphone stationed about 18 inches from the point of impact. The ratings are given in the same form as the transmitted noise, in that the intensity level was referred to that of the bare concrete at each frequency and the average of the differences for the 6 frequencies was taken as the rating of the floor. It will be noted that the wood floors were noisier than the concrete itself. It should be pointed out that some of the values which show reductions of around 5 decibels should probably have shown a greater reduction value, due to the interference of the mechanical noise of the tapping apparatus itself.

The data shown in the two tables together with data on other experimental panels tested, is shown graphically in Fig. 3, to illustrate the lack of correlation between values obtained above and below the floor. The connected points represent the reductions in noise transmitted by the various panels, plotted in the order of their effectiveness. The unconnected points show the differences in noise measured above the floor for the same panels.

The method which was used for obtaining these results was worked out for the practical purpose of determining the relative efficiency of several proposed types of floors, as compared to standard construction. Although we recognize deficiencies and faults which exist, the data obtained by this method gives good correlation with two other sets of tests made in previous years under different working conditions.

The general advantages may be stated: (1) Separate data is obtained on the transmitting *versus* the emitting properties. (2) There is no need to standardize variables of force or frequency of impact, microphone position, acoustical properties of rooms, etc., but only to maintain these constant during each set of tests. (3) The data is given in a form which is analogous to the transmission loss of floors and partitions for air-borne sound, and which has the same advantages. (4) The numerical rating obtained shows good correlation with the ear judgments of an observer.

Disadvantages of this method of testing are: (1) It is comparative rather than absolute and therefore one must test the standard floor every time a group of unknown floors are tested. (2) It is still necessary to determine just how satisfactory the standard concrete floor is under actual conditions of use in order to predict the performance of other floors.

There are several questionable assumptions: (*A*) Is the transmission of the unknown floor in relation to a standard floor independent of the area of either floor? Also, is it independent of the

hardness, weight, shape and frequency of the object making the impact? (*B*) If the hammer were faced with a softer material than the steel which was used, thus probably transmitting less high frequency components, would the method of computing the average reduction, as described above, be an accurate one which would give good correlation with the observer's ear method of rating?

This is a new field in which there is little data and absolutely no standardization of measurement technique. Just as a considerable amount of discussion has been presented and is still taking place on the proper method of measuring the sound absorbing characteristics of an acoustical material, it will probably involve considerably more experimenting by a number of observers before any standardization in this field is obtained. This paper is therefore written in the hope of stimulating interest in the problem.

The data listed above give relative values for common materials as obtained in these tests. Individual materials and constructions of the same general type are subject to considerable variation, and the test values shown should be interpreted accordingly.

Selective Amplification in Hearing Aids

N. A. Watson and V. O. Knudsen
University of California at Los Angeles, Los Angeles, California
(Received January 22, 1940)

I. Introduction

"SELECTIVE amplification" has become a by-word in hearing aid terminology and great claims have been made for it, yet many who have used the words and made the claims have understood very little about its fundamental principles or its application to hearing aids. The idea of using selective amplification in hearing aids has been known for many years, but the claims made for it are supported by meager quantitative data. The present investigation was undertaken to clarify the basic concepts of selective amplification in hearing aids; to obtain quantitative data for determining the effectiveness of selective amplification so used; and, on the basis of the data, to suggest tentative criteria which specify not only *whether* it should be prescribed, but also *how* best to prescribe it in cases which justify its use.

This first paper will be restricted to the results and prescription criteria for monaural listening by air conduction. Subsequent papers will deal with (1) air conduction listening with two ears, using first a single channel system then a true binaural (or two-channel) system; (2) various types of listening by bone conduction; and (3) criteria for determining whether air or bone conduction should be used in the prescription of a hearing aid. In this paper, therefore, unless another means of listening is specified, monaural air conduction listening is implied.

II. Basic Principles of Selective Amplification in Hearing Aids

At the outset it is essential that certain terms used in this paper be defined.

Amplification is the increase in level (measured in decibels) brought about by any part or all of the amplifying system. It may refer to the increase for a single pure tone, or for as complex a combination of sounds as speech.

Over-all amplification is the increase in intensity level brought about by the hearing aid as a whole—from microphone to earphone in position on the listener's ear (or its equivalent).

Uniform amplification is the equal amplification of all frequencies in the speech range. It is *not* amplification of all speech frequencies to the same level; it is, rather, amplification of each speech frequency *by the same amount.*

Selective amplification, on the other hand, is amplification which varies in amount at different speech frequencies; further, it is defined here as a *controlled* variation from uniform amplification and *not* an *un*controlled or only *partially* controlled variation, arising from the natural frequency characteristics of the mechanical parts of the hearing aid.

It is also essential to recognize that the effectiveness of *each* type of amplification may be fundamentally limited by the hearing loss of the individual and that there will be only a restricted range within which the *type* of amplification can be varied with comfort. This fact has often been overlooked, or if recognized, misinterpreted.

The normal auditory sensation area for monaural listening with an air conduction receiver has been determined accurately. Within it, at the approximate frequencies and intensities indicated in Fig. 1, lie the various speech sounds as estimated by Steinberg in 1928.[1] Later researches at the Bell Telephone Laboratories by Steinberg and others have indicated that the threshold curves and the speech sound areas should be modified slightly, but the general estimates shown in Fig. 1 are still valid.

For an ear with a typical *conductive* impairment the auditory sensation area is approximately the same size as for the normal ear, but with both threshold of hearing and threshold of feeling raised—usually more at low than at high frequencies. For such an ear one would expect that uniform amplification could be used effectively, all speech sounds being amplified equally

[1] J. C. Steinberg, Trans. Soc. Mot. Pict. Eng. **12**, 633 (1928), Fig. 4 (reprinted in Bell Tel. Lab. Reprint B-377, Feb., 1929).

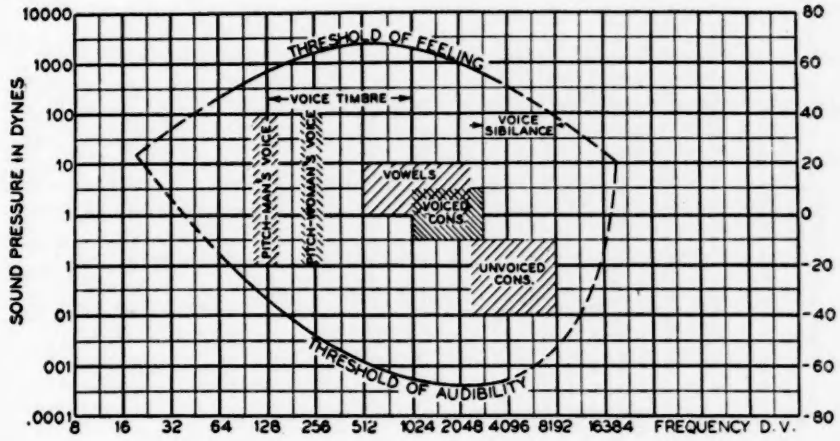

FIG. 1. Normal auditory sensation area for listening through an air conduction earphone. The approximate "areas" for various groups of speech sounds are indicated. (Steinberg.)

and brought to approximately the same positions in the conductive auditory sensation area as they occupied in the normal area. Further, one would expect that selective amplification would give little, if any, additional help.

For an ear with a typical *perceptive* impairment, however, the situation is quite different. The threshold of hearing curve is raised markedly, and usually more at the higher frequencies than at the low, while the threshold of feeling curve remains almost the same as for the normal or even is lowered slightly. The auditory sensation area is therefore smaller than normal and often is greatly distorted. It is impossible to bring all speech sounds from their usual positions of frequency and intensity into this area by uniform amplification. Either the low frequency components are made too loud, if the unvoiced consonants are brought within the area, or the unvoiced consonants are left out if the vowels are brought to a comfortable listening level within the area. In this type of impairment all of the sounds can be brought within the small distorted area, if at all, *only* by *selective* amplification, and, more specifically, by selective amplification which raises the high frequencies more than the low. But when this is done, the various speech sounds no longer have their usual relative levels. Therefore, for such a case it is impossible to make speech entirely audible and at the same time natural-sounding. It is to be expected, therefore, that a process of adaptation or accommodation to selective amplification may be necessary for a person with a marked perceptive impairment.

For an ear with a perceptive impairment so extreme that the upper frequency limit is lower than approximately 8000 cycles, it is *impossible* to bring all of the speech sounds into the distorted auditory sensation area because some of the consonants have characteristic frequencies which are only slightly below 8000 cycles and which are necessary for distinguishing the consonants one from another.

These considerations have shown that the speech sounds *must* be placed within the auditory sensation area to be *heard*. They have indicated that in some cases *all* of the speech sounds *may* be thus placed by the correct amplification and that in other cases, *most* of the sounds may thus be brought within the area. But they have not specified exactly *which* intensities of the sounds will be most effective. In determining these intensities, one must keep in mind that the equal loudness curves for ears with perceptive impairments, or mixed impairments with appreciable perceptive involvement, usually converge toward the threshold curve at the higher frequencies more than do the equal loudness curves for ears with normal hearing. (This convergence for impaired ears is indicated for several observers in the lower portions of Figs. 5, 6, 7, 8, 9, and 11, which will be discussed later.)

III. Criterion, Articulation Lists, Laboratory Apparatus, Technique of Testing

The criterion adopted for evaluating the effectiveness of the various types of amplification is based on the percentage syllable articulation, calculated according to the formulas developed by Fletcher and Steinberg.[2]

As the basis for the articulation lists to be used in these tests, a group of words was prepared by L. W. Sepmeyer, modeled after that compiled by Fletcher and Steinberg.[2] It contains sixty-nine words. Of these, twenty are "vowel" words, i.e., one-syllable English words each incorporating an English vowel sound between the consonants *b* and *t* or *b* and *k*; the remaining forty-nine are "consonant" words, i.e., one-syllable English words each containing an initial consonant before short *e* or long *i* or a final consonant after short *i*. Table I gives a typical seventy-five-word articulation list with the phrases used to introduce each group of three words. The sound to be tested by each word is italicized.

Table I. *Typical articulation list. The sound to be tested in each word is italicized. Words are called in groups of three, each group preceded by an introductory phrase.*

1. The first is	b*ait*	s*et*	r*et*
2. Now try	b*ut*	ri*p*	*ben*
3. Please repeat	b*oat*	s*ty*	ri*ff*
4. Group 4 is	b*ake*	g*uy*	w*in*
5. The next is	b*alk*	*jen*	*th*igh
6. Now comes	b*at*	*pen*	*yen*
7. Listen to	b*ite*	ri*m*	*let*
8. Try to hear	b*et*	*men*	ri*ng*
9. Next comes	b*ought*	*tie*	*thy*
10. Group 10 is	b*ook*	ri*ck*	w*ing*
11. Please repeat	b*it*	*shy*	w*ick*
12. Can you hear	b*out*	w*ry*	*jet*
13. Now try	b*eat*	*high*	*yet*
14. The next is	b*oot*	*net*	*wen*
15. Listen to	b*eck*	*vet*	w*ill*
16. Try to hear	b*oot*	*hen*	*sigh*
17. The next is	b*ook*	*fen*	ri*g*
18. Now comes	b*ike*	wh*iz*	*pet*
19. Now try	b*ack*	*then*	ri*d*
20. Group 20 is	b*it*	ri*ch*	w*ith*
21. Listen to	b*out*	ri*b*	*ten*
22. Try to hear	b*eak*	*fie*	w*ish*
23. Please repeat	b*uck*	wh*iz*	wh*ist*
24. Next comes	b*oat*	wh*ich*	*den*
25. The last is	b*ark*	*vie*	*wet*

In order to insure a source of these speech sounds of standard, accurately reproducible intensity and quality, phonograph recordings of sixteen different arrangements of the test words were made under the supervision of Mr. Sepmeyer. The recording system was accurately calibrated as to frequency response. The phonograph-amplifier system used to reproduce the recordings has a frequency response curve such that when the recordings are played over it, the words reach the listener uniformly* amplified or reduced from the level at which they were originally spoken, except when controlled variations are introduced into the amplifier system to provide selective amplification. An extensive set of frequency distortion networks is used for this purpose.

The listener, or observer, is situated in a room[3] which is so nearly sound-proof that he can hear nothing but the test sounds coming through the earphones. This is true even for observers with normal hearing.

The threshold level for speech for each ear is determined by the observer listening to the articulation lists in the following manner: The operator controls a noiseless switching device so that only the test words are heard in the earphone, the introductory phrases being cut out in each case. The observer is instructed to signal for each word he hears. The operator first sets the attenuators at a level such that the observer can hear all three words faintly; then he decreases the intensity by $1\frac{1}{2}$ db after each group of three words until the observer can no longer hear any of the words. A complete record of the response or no-response for each word is kept. The operator then increases the intensity by $1\frac{1}{2}$-db steps, until an intensity is reached at which the observer hears all three words. The process is then repeated for the remainder of the list, so that a total of seventy-five words is used to determine each threshold. Usually this means that approximately 15 words are presented to the observer at each level close to the threshold. From these results the threshold level is chosen as the lowest level at which the observer responds to more than half of the words presented at that level. Tests are then conducted at various levels above threshold, obtained by removing attenuation from the amplifying system. As the listener-observer hears the successive groups of three words, he writes them on a special form. The list is then graded and the

[2] H. Fletcher and J. C. Steinberg, "Articulation Testing Methods," Bell Sys. Tech. J. **8**, 806 (1929); reprinted in Suppl. to the Jan., 1930, J. Acous. Soc. Am., pp. 1–97.

* Within ±3 db from 100 to 7000 cycles.

[3] The details of the laboratory are given in J. Acous. Soc. Am. **9**, 99 (1937), Fig. 4.

percentage syllable articulation calculated from the correct Fletcher-Steinberg formula[2] for the type of list given in Table I; namely,

$$S=\left\{1-\left[1-\frac{V}{100}\left(\frac{C}{100}\right)^{2}\right]^{0.9}\right\}100,$$

where S is the percentage syllable articulation, V is the percentage of vowels correctly interpreted in the vowel words, and C is the percentage of consonants correctly interpreted in the consonant words.

IV. Results with Uniform Amplification

1. Increase of percentage syllable articulation with increase of intensity level

The increase of percentage syllable articulation with increase of intensity level has been studied for normal-hearing observers by Fletcher[4] and by Steinberg[5] and for a limited number of observers with impaired hearing by Steinberg and Gardner[6, ‡] and by Kelley.[7] In the present investigation this increase has been carefully determined for hard-of-hearing observers with widely different types of impairment. Representative curves are shown in Fig. 2;* for comparison, the normal curve as given by Steinberg is included.

The curves for observer 10, left ear, and observer 13, left ear, rise *less* steeply than does the curve for the normal over most of the range of levels; the curves for observer 7, right ear and left ear, rise *more* steeply than does the normal curve over part of the range of levels. This indicates immediately that the hearing of speech by impaired ears listening with uniform amplification at above-threshold levels cannot, in general, be predicted from the difference between the normal ear threshold level for speech and the impaired ear threshold level for speech. This difference will hereafter be called the *hearing loss for speech at threshold*, or more briefly, *loss at threshold*.

For example, consider the curve of observer 10, left ear. This ear has a threshold level for speech which is 65 db higher than that for the normal ear. Thus at zero percentage syllable articulation, which corresponds to threshold, there is a 65-db horizontal separation between observer 10's left ear curve and the normal ear curve, i.e., a 65-db hearing loss for speech at threshold. At the percentage syllable articulation which corresponds to observer 10's most comfortable level for listening to speech† with his left ear (40 db above his left ear threshold for speech and 105 db above the normal ear threshold for speech) there is a horizontal separation of 78 db between the curve for his left ear and the curve for the normal ear. This horizontal separation will hereafter be referred to as the *hearing loss for speech at the most comfortable level*, or, *loss at most comfortable level*. Thus observer 10, left ear, has a 78-db loss at his most comfortable level as compared to a 65-db loss at threshold. (The air and bone conduction threshold curves—for pure tones—for his left ear are shown in the lower part of Fig. 5.)

On the other hand, for observer 7, left ear, the loss at threshold is 77 db (12 db greater than for observer 10, left ear) yet the loss at his most comfortable level (23 db above his threshold for speech and 100 db above the normal ear threshold for speech) is only 66 db (as compared to observer 10's loss of 78 at the most comfortable level for his left ear). Moreover, the maximal percentage attained by observer 7, left ear, is 89, which is 29 percent greater than that attained by observer 10, left ear. (The air and bone conduction threshold curves for observer 7, left ear, are given in Fig. 6.)

An even more extreme contrast is given by the curves for observer 13, left ear, and observer

[4] H. Fletcher, *Speech and Hearing* (D. Van Nostrand, New York, 1929), Fig. 127, p. 272.

[5] J. C. Steinberg, J. Acous. Soc. Am. **1**, 121 (1929), Fig. 2.

[6] J. C. Steinberg and M. B. Gardner, J. Acous. Soc. Am. **10**, 255 (1939).

‡ After the present paper was sent to press, the complete article by Steinberg and Gardner (the reference for the abstract of which is given in footnote 6) appeared in J. Acous. Soc. Am. **11**, 270 (1940).

[7] N. H. Kelley, Arch. Otolaryngology **29**, 506 (1939) and J. Exper. Psychol. **21**, 342 (1937).

* As an example of the extent of data obtained for the preparation of these curves: 85 lists such as are illustrated in Table I were listened to by observer 10 to determine his left-ear curve of Fig. 2.

† This most comfortable level for listening to speech, or more briefly, *most comfortable level*, is determined by raising and lowering the intensity of the speech until the observer indicates a certain level as most comfortable. For some observers the level chosen is quite close to the highest tolerable level; for others it is considerably below the upper limit. In all cases the observer's judgment is accepted as final.

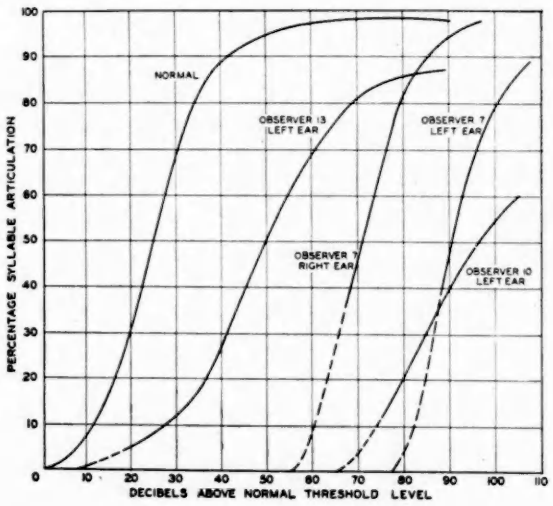

FIG. 2. Increase of percentage syllable articulation with increasing level above normal threshold: for the standard normal ear (as given by Steinberg), and for the ears of observers with widely different impairments of hearing (tested in the present investigation).

7, right ear (also shown in Fig. 2). Observer 13, left ear, has a loss for speech at threshold of only 8 db, yet the loss at his most comfortable level, (60 db above his threshold) is 36 db, which is 28 db *greater* than at threshold. (His left-ear air and bone conduction threshold curves are given in Fig. 7.) Observer 7, right ear, with a loss at threshold of 55 db (as compared to observer 13's 8 db for his left ear) has a loss at his most comfortable level (30 db above his threshold) of only 45 db which is 10 db *less* than at threshold. Moreover, in spite of having a loss at threshold 47 db greater than observer 13, left ear, he attained a maximal articulation of 97 percent, which is 10 percent greater than the maximal 87 percent, attained by observer 13. (The air and bone conduction threshold curves for observer 7, right ear, are given in Fig. 8.)

The variations of the curves for impaired ears from the curve for the normal ear, as given in Fig. 2, may be made more conspicuous by replotting the data of Fig. 2 as in Fig. 3. In the latter figure, the abscissae are the levels above the impaired ear threshold and the ordinates on the left are levels above the normal ear threshold. The ordinates on the right give the percentage articulations attained by the normal ear at levels taken at intervals of 10 db, starting at threshold. These percentages are derived from the normal curve in Fig. 2, to which reference should be made for exact percentages at intermediate levels. For each level above threshold for a given impaired ear a certain percentage articulation is attained (as shown in Fig. 2); the same percentage is attained by the normal ear at a level above normal threshold which may be the same numerically—or may be different. In either case, the two levels, one for the impaired ear and one for the normal, determines one point on the comparison curve between the impaired ear and the normal ear as plotted in Fig. 3. Similarly other pairs of levels determine other points, and through them the comparison curve may be drawn. If the increase of percentage articulation with level above threshold were the same for the impaired ear as for the normal, the two values in each pair of levels above threshold would be equal and the comparison curve would be a straight line, starting at the origin and bisecting the quadrant at 45° to each axis. This is not the case for all impaired ears, however, as has been indicated in Fig. 2 in which the curves have different slopes. In Fig. 3, the "comparison-to-normal" curves are given for observer 10, left ear; observer 13, right ear; observer 13, left ear; observer 7, right ear; and observer 7, left ear. It is apparent that the curve for observer 13,

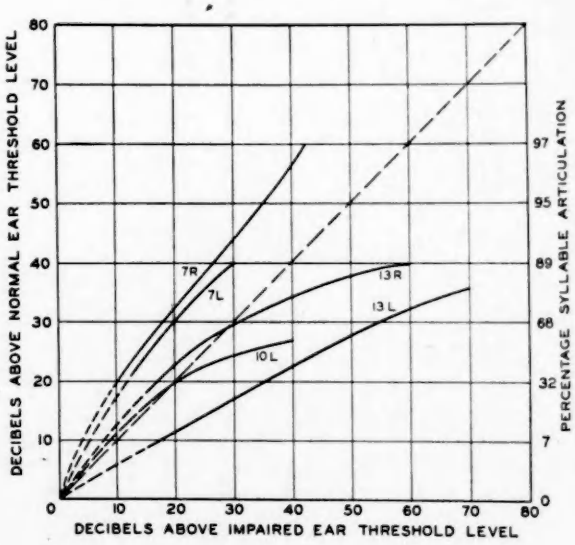

FIG. 3. Comparison curves between the normal ear and ears with widely different types of impairment as to the ways in which their percentage syllable articulation increases with increasing level above their respective thresholds (based on the data of Fig. 2). See the text for the method of construction of the comparison curves.

left ear, lies below the 45° line for all levels and increasingly far below for each higher level; this corresponds to observer 13's left ear curve* being horizontally separated from the normal curve more at all levels *above* threshold that *at* threshold, and rising less steeply than the normal ear curve. Similarly the portions of the other curves which lie below the 45° line in Fig. 3 represent portions of the impaired ear curves of Fig. 2 which have greater horizontal separation from the normal curve at the corresponding levels above threshold than at threshold. The portions of the other curves which lie above the 45° line represent portions of the impaired ear curves of Fig. 2 which have smaller horizontal separations from the normal curve at corresponding levels above threshold than at threshold, and which rise more steeply (at least for a portion of the curve) than the normal curve does.

Further comparisons between the hearing loss for speech at threshold and the loss at most comfortable level for the impaired ear are given in Table II. The first seven ears listed have perceptive losses or mixed losses with the perceptive loss, as estimated from the air and bone conduction threshold curves, predominant. For all seven ears the loss at most comfortable level is greater than the corresponding loss at threshold. The next seven ears in Table II have losses which are conductive or mixed, with conductive predominant. For all seven ears the loss at threshold is almost the same as that at the corresponding most comfortable level above threshold. (The greatest variation is 4 db.) In the third group are two ears with mixed losses, but with the conductive predominant, and

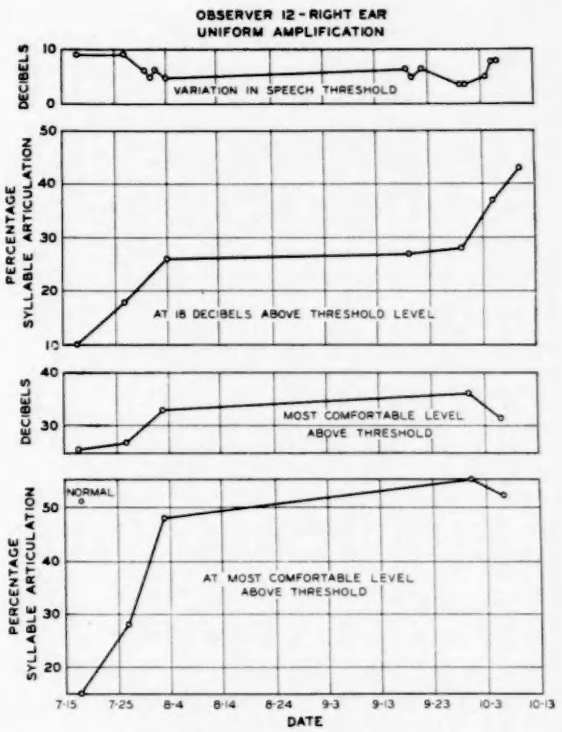

FIG. 4. Accommodation effects for a single observer—uniform amplification.

* In Fig. 2.

TABLE II. *Comparison between the hearing loss for speech at threshold and the hearing loss for speech at most comfortable level for observers with widely different types of impairment.*

OBSERVER AND EAR	TYPE OF LOSS	HEARING LOSS FOR SPEECH AT THRESHOLD (DECIBELS)	HEARING LOSS FOR SPEECH AT MOST COMFORTABLE LEVEL (DECIBELS)
12 Right	Pure perceptive	63	73
12 Left	Pure perceptive	76	89
13 Right	Perceptive	4	20
13 Left	Perceptive	8	36
10 Left	Mixed, perceptive predominant	65	78
17 Right	Beginning perceptive	15	20
5 Right	Primarily perceptive	18	25
5 Left	Primarily conductive	48	48
3 Right	Beginning conductive	15	11
3 Left	Beginning conductive	5	2
14 Right	Mixed, conductive predominant	44	44
14 Left	Conductive predominant	42	39
1 Right	Conductive	45	48
1 Left	Conductive	40	42
7 Right	Mixed, conductive predominant	55	45
7 Left	Mixed, conductive predominant	77	66

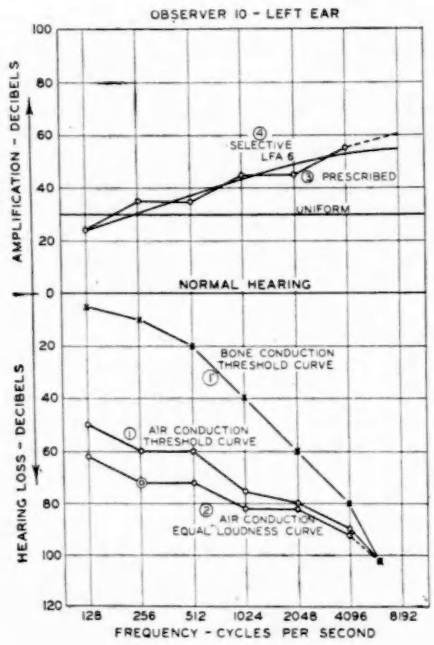

FIG. 5. Air and bone conduction threshold curves (1 and 1′) and the "most comfortable equal loudness curve" for air conduction (2) which are used to prescribe selective amplification for an individual ear; the prescribed amplification curve (3); and the actual amplifier curve approximating it (4).

diagnosed as otosclerosis. For these two ears the loss at most comfortable level is definitely less than that at threshold.

The two ears of observer 5 suggest an interesting comparison, because the impairment in the right ear is primarily perceptive (as estimated from the air and bone threshold curves, shown in Fig. 9) while that in the left is primarily conductive (Fig. 10). Since the mental processes beyond the two ears are the same, the differences shown must be due to the difference in ears (including nerves). The left fits into the conductive group, according to the data in Table II, and the right into the perceptive group.

2. Accommodation effect with uniform amplification

Some observers, particularly those with a large perceptive loss, do not hear well with uniform amplification when it is first tried, because they have never heard some of the speech sounds before, or have not heard them for a long time. Such an observer was No. 12, with a severe perceptive loss at all frequencies. He scored a syllable articulation of only 10 percent when he first listened with his right ear to uniformly amplified speech at a level of 18 db above his threshold for speech. (This was 5–10 db below his most comfortable level.) He said it sounded very unnatural. On the basis of his right-ear threshold curves as shown in Fig. 11, and results with other observers with similar losses, this percentage seemed much too low. Accordingly, a series of tests was inaugurated to determine whether he could accommodate himself to uniform amplification. The results of these tests, which were continued over a period of three months, are given in Fig. 4. The threshold for speech varied slightly, but not significantly. The articulation at a level 18 db above threshold increased markedly, from 10 to 26 percent, during the first three weeks of testing. This improvement cannot be attributed to observer 12's degree of interest and cooperation, or to an increasing familiarity with the lists; he was intelligent, cooperative, and thoroughly acquainted with the content of the lists before he started the series of tests. It was undoubtedly an accommodation to the hearing of *new* sounds. After a lapse of about six weeks the tests were resumed. The percentage syllable articulation was about the same as when the tests were interrupted. It increased for further tests until it reached a maximum of approximately 43 percent. The

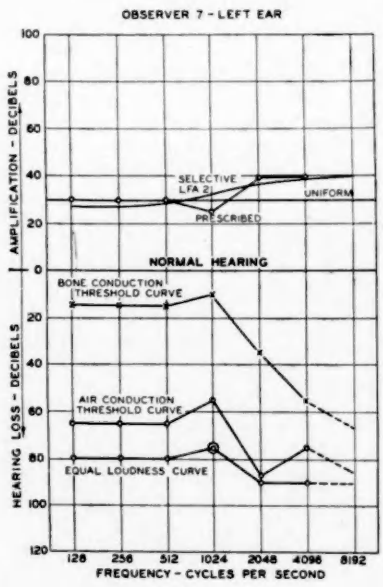

FIG. 6.

most comfortable level for listening to speech with uniform amplification also increased, considerably at first, from 26 to 33 db, but then it leveled off and fluctuated slightly around 33 db thereafter. The syllable articulation for his most comfortable level, which at the beginning of the tests was only 15 percent, also increased as the tests were continued, reaching a maximum of about 55 percent. His accommodation to selective amplification will be discussed later.

V. Methods of Prescribing Selective Amplification

In experiments on hearing, the observer's judgment is accepted concerning many things: threshold, most comfortable level, equal loudness level, etc. In such matters his judgment *must* be accepted; in most instances his consistency indicates that this judgment is surprisingly accurate. When it comes to the question, "With which kind of amplification do you hear best?"

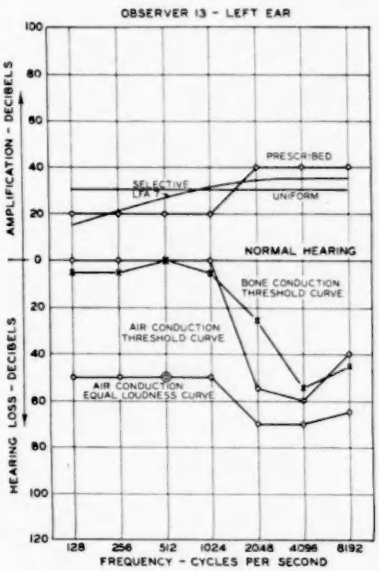

Fig. 7.

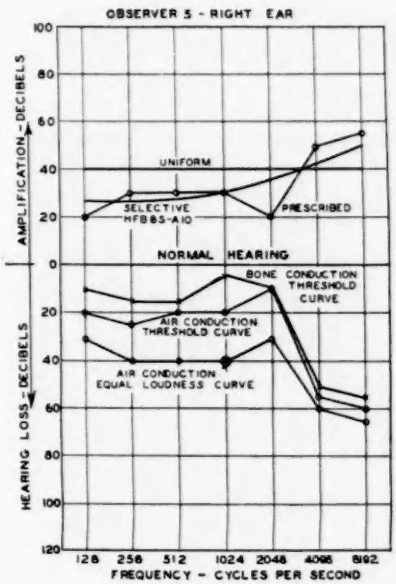

Fig. 9.

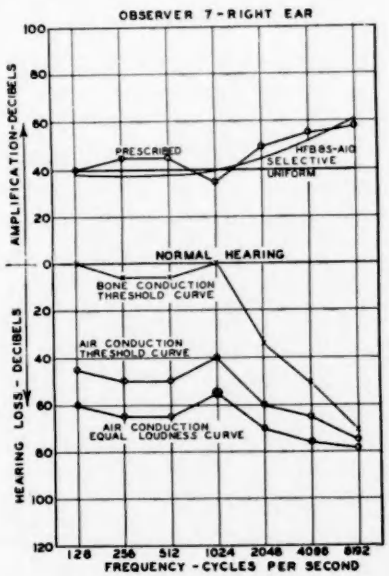

Fig. 8.

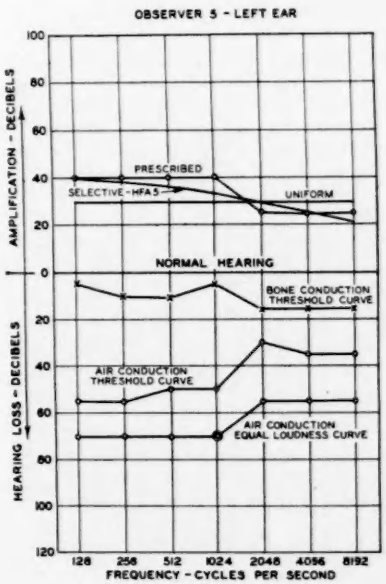

Fig. 10.

the experience of the authors indicates that his judgment is not reliable—even when he conscientiously sets himself against choosing the one he "likes" best. Until he becomes accommodated to the proper amplification he almost invariably chooses as "best" a "peaked" amplifier or one which accentuates most the frequencies he already can hear best; and, paradoxically, he frequently says he hears poorly with the amplifier with which he actually hears best. For example, observer 12, with a loss in his right ear which was greater for high frequencies than for low, as shown in Fig. 11, chose a 1000-cycle peak amplifier or selective amplifier HFA-5 (which amplifies the high frequencies less than the low ones by the amount indicated in the upper part of Fig. 10), in preference to a uniform amplifier or selective amplifier LFA-7 (which amplifies the high frequencies more than the low by the amount indicated in Fig. 11). Yet either of the latter two (Uniform or LFA-7) allowed him to hear better than either of the two he chose as "best." The results for this observer at his most comfortable level for each filter (33 db above the respective threshold in each case) are given in Table III.* Results are also given for other observers. In each case the type or types of amplification chosen as "best" by the observer are starred. On the basis of these quantitative results, the conclusion is obvious that any method of prescription which depends on the observer's direct judgment of what seems to him to be the best type of amplification is extremely unreliable. This has been recognized by many investigators, yet most hearing aids are prescribed, in practice, by methods which rely on just such judgments by prospective users.

TABLE III. *Comparison (for three observers) between the effectiveness of amplifiers chosen as "best" by the observers and amplifiers providing uniform amplification or amplification correctly prescribed. The amplifiers chosen as "best" by the observers are starred.*

OBSERVER AND EAR	TYPE OF AMPLIFIER	LEVEL ABOVE THRESHOLD (DECIBELS)	PERCENTAGE SYLLABLE ARTICULATION
12 Right	*1000-cycle peak	33	38
	*HFA 5	33	47
	Uniform	33	50
	LFA 7	33	61
10 Left	*HFA 5	30	38
	Uniform	30	48
	LFA 6	30	72
13 Left	*HFA 5	60	70
	Uniform	60	80
	LFA 7	60	87

* Each percentage given in Table III, and also in Tables IV and V, is based on at least 375 test words.

In recent years, several methods of prescribing selective amplification have been proposed which were based in one way or another on the hearing losses at representative frequencies in the speech range—i.e., on the audiogram. Some of the methods are absurd; others are of sufficient worth to justify a discussion of their advantages and deficiencies.

One such method prescribes an amplification at each frequency which is equal to the hearing loss at that frequency, or more conservatively, amplification equal to the hearing loss minus a constant, the constant being the same for each frequency. This method is not always satisfactory as the following considerations show. Assume a prescription has been made on this basis for a person with severe perceptive hearing loss, much greater at high than at low frequencies. An amplifier fulfilling the prescription and bringing the fundamental frequencies of the voice and the lower frequency components of the vowels to a comfortable listening level would amplify the higher frequency components of the vowels and some of the consonants too much, making them painfully loud. This is in sharp contrast with the results for uniform amplification for such a case, as uniform amplification leaves the higher frequency components of the vowels and the consonants too faint when the lower frequency components of the vowels are adjusted to a comfortable level.

Another general method plots the auditory sensation area for the impaired ear, using the same type of chart as in Fig. 1. It then sets up an amplifier curve having a shape which lies within the distorted auditory sensation area. One special form of this method suggests bisecting each ordinate between the threshold of hearing and the threshold of feeling curves and joining the points thus determined to form "the optimum shape of the amplifier characteristic required." That such an amplifier curve is optimal does not follow. Simply because the amplifier curve has a shape which permits it to be fitted into the

auditory sensation area does not insure that the amplifier will bring the various speech sounds to the proper levels. As a matter of fact, a curve so fitted oftentimes does not; for example, if it is prescribed for an ear with one particular perceptive loss the curve is nearly a straight line and therefore essentially uniform amplification, which is unsatisfactory as discussed above. Similarly all other criteria based solely on fitting an amplifier curve into the auditory sensation area give no assurance of success. Their proponents have made the mistake of trying to fit the *amplifier curve* into the distorted auditory sensation area rather than to fit the *frequency-intensity areas* occupied by the amplified speech sounds into the diminished and distorted auditory sensation area.

The failure of these methods, based on the threshold of hearing curve alone, or even on the thresholds of hearing and feeling, emphasizes the necessity, as stated above, of taking into account the convergence of equal loudness contours toward the threshold curve, especially for an ear with a perceptive impairment.

The recognition of these facts during the early stages of this investigation suggested what the authors consider to be a more rational criterion for prescribing selective amplification; namely, a criterion based on a "most comfortable equal loudness curve." With slight modifications, this criterion has proved the most effective one the authors have been able to devise, for the observers thus far tested. The criterion and the technique for determining the proper amplifier to comply with it are as follows (application to the left ear of observer 10 is used as an example).

(1) Determine and chart the air and bone conduction threshold curves. (Curves 1 and 1′ in Fig. 5.)

(2) Determine and chart the "most comfortable equal loudness curve" for air conduction as follows: Use 1000 cycles (or 1024 cycles) for the reference tone if the bone conduction loss at that frequency is as small as or smaller than at lower frequencies; if not, use 500 (or 512) cycles or 250 (or 256) cycles, choosing the frequency at which the ear shows the smaller bone conduction loss. Find the most comfortable listening level for the chosen reference tone. Determine for frequencies above and below the reference tone,

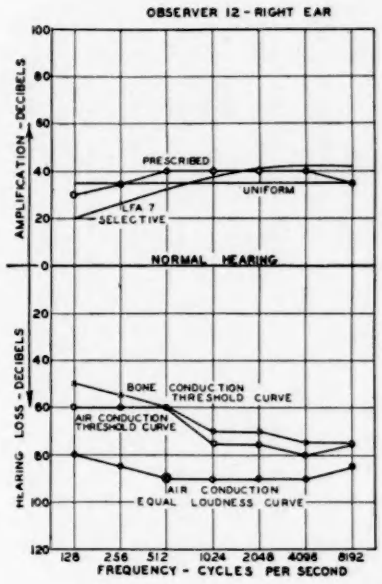

FIG. 11.

at octave intervals, the levels which sound equally as loud to the observer as the reference tone does at the most comfortable level. (For observer 10, left ear, 256 cycles was chosen as the reference frequency, because the bone conduction hearing loss was 40 db at 1024 cycles, 20 db at 512 cycles, and only 10 db at 256 cycles. The most comfortable level at 256 cycles was found to be 12 db above the threshold for this ear. The "most comfortable equal loudness curve" determined on the basis of this reference frequency and intensity is curve 2 in Fig. 5.)

(3) Calculate the amplification for each frequency (curve 3) from the formula

$$A_F=[HL_F-(D_{RF}-D_F)+K]$$

where A_F is the amplification (in decibels) to be prescribed at the frequency F; HL_F is the hearing loss (in decibels) at that frequency; D_{RF} is the difference (in decibels) at the reference frequency between the air conduction hearing loss curve (No. 1) and the most comfortable air conduction equal loudness curve (No. 2); and D_F is the difference (in decibels) at the frequency F between curves 1 and 2. $(D_{RF}-D_F)$ thus is a measure of the convergence of curves 1 and 2. $[HL_F-(D_{RF}-D_F)]$ gives the shape of the prescribed amplification curve (No. 3), which is simply the mirror image of curve 2 over the zero line. K is a constant which is added

to or subtracted from the amplification at all frequencies to adjust the intensity to the most satisfactory value for various listening conditions. Its maximal value is set by the position of the observer's threshold of feeling. In practice it must be determined by the hearing aid user through his adjustment of the volume control to give tolerable listening conditions. (For observer 10, left ear, the prescribed curve is given by curve 3, with the amplification set by curves 1 and 2 and an arbitrary value of $K = -37$ db.)

In Fig. 5, curve 4 is the actual amplifier curve (LFA-6) chosen to approximate the prescription for observer 10's left ear. It is shown compared to the uniform amplification curve which is arbitrarily placed at 30 db.

Selective amplifier curves were prescribed for the other observers in the same manner. Examples of them and the "most comfortable equal loudness curves" from which they were determined are given in Figs. 6, 7, 8, 9, 10 and 11. In each case the reference frequency and most comfortable level for that frequency are indicated by a large circle drawn around the corresponding point of the "most comfortable equal loudness curve."

VI. Results with Selective Amplification

In some instances properly prescribed selective amplification proved superior to uniform when first tried; in other instances uniform amplification proved more effective. This latter was true for observer 12, right ear, as indicated in Fig. 12 for a level above threshold of 18 db. However, an accommodation effect took place for selective amplification which was greater than that for uniform, so that the selective was, within a few weeks, considerably more effective than uniform and remained so in spite of the later large accommodation effect with uniform. This was also true at higher levels. At a level of 33 db above threshold, for example, the syllable articulation for uniform was 50 percent and for selective LFA-7, 61 percent.

Detailed results, showing the effectiveness of selective amplification for a single observer, are plotted in Fig. 13 for observer 10, left ear, using amplifier LFA-6, which approximates the prescription proposed for that ear. At all but very

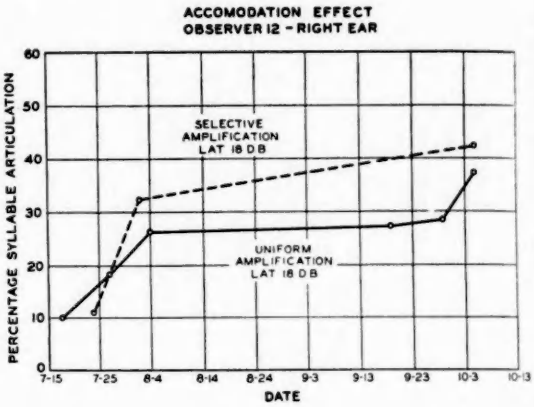

FIG. 12. Accommodation effects for a single observer—uniform amplification and selective amplification, each at a level above threshold of 18 db.

small levels above his threshold, the selective was superior to uniform. At 30 db above his threshold, selective amplification gave a syllable articulation of 72 percent, which is 24 percent greater than for uniform for the same ear and approximately equal to (actually 3 percent greater than) that for a normal ear using uniform amplification. Seventy-two percent is also very close to the maximal percentage of 75 possible for observer 10 using the lists of the type in Table I. This value for the maximal possible percentage is based on the following facts: (1) the upper frequency limit for observer 10, left ear, is 6000 cycles; and (2) several consonants (*s*, *z*, *f*, *v*, voiced *th*, and unvoiced *th*) have characteristic frequencies above 6000 cycles which are necessary for sure recognition of them. Thus observer 10 can never identify these sounds with certainty with his left ear. For the lists used, errors on these sounds will reduce the average percentage syllable articulation to 75.

If the curve for selective amplification for observer 10's left ear, as shown in Fig. 13, is compared to the normal ear curve for uniform amplification, and a comparison curve of the form given in Fig. 3 is constructed, it will be very close to the 45° line of Fig. 3. The *shift* from the 10L curve shown is a measure of the superiority of the selective amplification for observer 10, left ear. Similarly the use of prescribed selective amplification with observer 13, right ear, brings the articulation for observer 13's right ear at a level of 60 db above its threshold up to the percentage attained by the normal ear at the

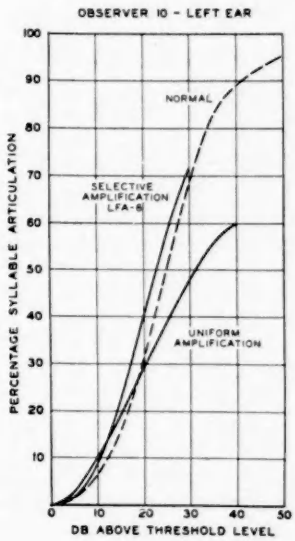

FIG. 13. Increase of percentage syllable articulation with increasing level above threshold for a single impaired ear using (1) uniform amplification and (2) selective amplification prescribed by the criterion based on the "most comfortable equal loudness curve" as in Fig. 5; the increase for the standard normal ear (as given by Steinberg) is shown for comparison.

same level above its threshold. In Fig. 3 this means bringing curve 13*R* up to the 45° line at 60 db. The use of prescribed selective amplification with observer 13, left ear, brings the articulation up some, but not to normal. This probably is due to this observer's large and extremely abrupt loss at high frequencies for the left ear as is indicated in Fig. 7. Thus his 13*L* curve of Fig. 3 would be brought up, but not quite to the 45° line. The use of prescribed selective amplification with observer 7 proved beneficial for both ears—allowing articulations even higher than with uniform amplification for the same levels. In Fig. 3 this would mean the swinging of curves 7*R* and 7*L* even higher than they are for uniform amplification.

Table IV gives the results for a number of observers, using first uniform amplification and then selective amplification, prescribed according to the criterion of the most comfortable equal loudness curve. From this table it appears that selective amplification so prescribed helps almost every impaired ear more than does uniform amplification at the same level above threshold. That the particular type of selective amplification prescribed by the most comfortable equal loudness curve is superior to others has been demonstrated by trying selective amplifiers with frequency characteristics on either side of the prescribed amplifier. Results for several observers are given in Table V, in which the prescribed amplifiers are starred. For example: for observer 7, left ear, the prescribed amplification was LFA-2, a type which amplifies sounds with frequencies near 8000 cycles approximately 13 db more than sounds with frequencies near 1000 cycles. (See Fig. 6.) The syllable articulation for observer 7, left ear, using this amplifier at a level of 18 db above his threshold was 79 percent. For uniform amplification at the same level above

TABLE IV. *Comparison between the hearing of speech by uniform amplification and by prescribed selective amplification, both at equal levels above threshold, by observers with widely different types of impairment.*

OBSERVER AND EAR	TYPE OF LOSS	LEVEL ABOVE THRESHOLD (DECIBELS)	PERCENTAGE SYLLABLE ARTICULATION FOR UNIFORM AMPLIFICATION	PRESCRIBED SELECTIVE AMPLIFIER	PERCENTAGE SYLLABLE ARTICULATION FOR PRESCRIBED SEL. AMPLIFICATION
12 Right	Pure perceptive	33	50	LFA 7	61
13 Right	Perceptive	60	91	LFA 6*a*	97
13 Left	Perceptive	60	80	LFA 7	87
10 Left	Mixed, perceptive predominant	30	48	LFA 6	72
16 Left	Beginning perceptive	30	57	HFB 8*S*, Att. 10	73
17 Right	Beginning perceptive	30	55	HFB 6*S*, Att. 10	58
5 Right	Primarily perceptive	30	78	HFB 8*S*, Att. 10	82
5 Left	Primarily conductive	30	82	HFA 5	86
3 Right	Beginning conductive	45	85	1000-cycle dip	92
3 Left	Beginning conductive	30	82	HFA 4*A*	80
14 Left	Conductive predominant	39	91	1000-cycle peak	96
1 Left	Conductive	36	77	LFA 4, Att. 10	78
7 Left	Mixed, conductive predominant	23	80	LFA 2	86
7 Right	Mixed, conductive predominant	30	91	HFB 8*S*, Att. 10	96

threshold it was 73 percent. For amplifier HFA-7*a*, which amplified sounds with frequencies near 100 cycles approximately 9 db more than sounds with frequencies near 8000 cycles, it was only 37 percent. Thus better results were obtained by amplifying the high frequencies somewhat more than the low. If the high frequencies were amplified too much, however, as by LFA-6, which amplifies sounds with frequencies near 8000 cycles almost 30 db more than those with frequencies near 100 cycles, the articulation (62 percent) was not so high as that for the prescribed amplification for the same level above threshold, namely, 79 percent.

As indicated by the results of Table IV, the superiority of properly prescribed selective amplification over uniform is greater for ears with perceptive impairment than for those with conductive impairment. As a matter of fact, the additional help from selective amplification for most of the cases with conductive impairment predominant is so small that it is questionable whether the trouble or cost of prescribing a selective amplifier is warranted. Thus it would seem to be desirable to have a criterion to determine *whether or not* to prescribe selective amplification—as well as *how* to prescribe it. Such a criterion (tentatively suggested on the basis of the data in Tables II and IV) with the method of applying it to an individual case is as follows.

(1) Determine the air and bone conduction threshold curves and the air conduction hearing loss for speech at threshold.

(2) Determine the most comfortable level above threshold for listening to speech; take several articulation lists at this level, and from these calculate the corresponding percentage syllable articulation; find the hearing loss for speech (in decibels) at this most comfortable level by comparison with the normal curve given in Fig. 2.

(3) If the hearing loss for speech at the most comfortable level, thus determined, is *approximately equal* to the hearing loss for speech at threshold, prescribe uniform amplification with the assurance that it will be of much help, and that selective amplification would be little, if any, more effective.

(4) If the hearing loss for speech at the most comfortable level is *less* than the hearing loss for speech at threshold, prescribe selective amplification by means of the criterion based on the "most comfortable equal loudness curve" with the expectation that it will be of as much help as uniform amplification, and in some cases *more*. (This statement is based on the results for two ears for which selective amplification was of more help than uniform amplification.)

(5) If the hearing loss for speech at the most comfortable level is *appreciably greater* than the hearing loss for speech at threshold, prescribe selective amplification by means of the criterion based on the "most comfortable equal loudness curve" with the assurance that it will give *as much* help as uniform amplification, *probably more*, and in some cases *much more*.

The procedure here recommended for the prescribing of hearing aids has not yet been tested

TABLE V. *Comparison (for seven observers) between the effectiveness of selective amplification prescribed according to the "most comfortable equal loudness curve" and other types of selective amplification with frequency characteristics lying on either side of that of the prescribed type. The prescribed selective amplifier is starred in each case.*

OBSERVER AND EAR	SELECTIVE AMPLIFIER	LEVEL ABOVE THRESHOLD (DECIBELS)	PERCENTAGE SYLLABLE ARTICULATION
12 Right	HFA 5	33	47
	*LFA 7	33	61
	LFA 6	33	56
13 Left	HFA 5	60	70
	Uniform	60	80
	*LFA 7	60	87
	LFA 6	60	86
13 Right	HFA 5	60	90
	Uniform	60	91
	LFA 7	60	95
	*LFA 6*a*	60	97
5 Left	*HFA 5	30	86
	Uniform	30	82
	HFB 8*S*, Att. 10	30	74
10 Left	HFA 5	30	38
	Uniform	30	48
	*LFA 6	30	72
7 Left	HFA 7*a*	18	37
	Uniform	18	73
	*LFA 2	18	79
	LFA 6	18	62
7 Right	HFA 7	30	80
	Uniform	30	91
	*HFB 8*S*, Att. 10	30	96
	LFA 6	30	85

sufficiently to advocate its use as a standard clinical technique. It is to be expected that further improvement, and possible simplification, will come from more extended tests. However, the experience of the authors to date indicates that this procedure marks a distinct improvement over the methods currently used.

This investigation has been supported by research funds of the Physics Department of the University of California at Los Angeles, and by generous grants in aid from the Board of Trustees of the Research Fund of the American Otological Society, Incorporated. Mr. L. W. Sepmeyer constructed much of the apparatus used, and has offered helpful advice during the course of the investigation. Mr. R. S. Gales constructed some of the apparatus, conducted most of the tests, and, in some instances, suggested the course of further testing. Mrs. N. A. Watson corrected all of the tests, charted the data, assisted in many of the tests, and rendered many helpful services in the preparation of the report. The many hard-of-hearing observers cooperated both patiently and commendably. To all these collaborators we acknowledge our indebtedness and express our gratitude.

Studies of Acquired and Inherited Deafness in Animals*

M. H. LURIE
Mosher Laboratory, Massachusetts Eye and Ear Infirmary, and Departments of Otology and Laryngology, Harvard Medical School

(Received January 20, 1940)

IN this address to you, the members of the Acoustical Society of America, I wish it to be understood that I come as an otologist with only a general knowledge of acoustics. Thus, I leave all the physical and engineering problems to you, for the otologist concerned with the living response to sound has quite a few problems of his own. The differentiation between the perception of sound and the physics of sound is an extremely important one, but neither, by itself, can solve the mystery of how we hear. They must combine to achieve any progress in the understanding of this vital and interesting phenomenon of hearing. The achievements and progress of the acoustical engineers and scientists have been extraordinary and especially rapid in the past ten years. It is to you men, who have helped to develop all this, that the otologist and physiologist are most grateful. For, with the development of the radio and apparatus that could detect and amplify the most minute electrical changes, a new method for the study of hearing became possible. This method, first used by Professors Wever and Bray in 1930,[1-2] has enabled us to study hearing in the living organism. The placing of an electrode on the auditory nerve and amplifying the electrical changes has enabled us to analyze the Wever and Bray phenomenon into its component parts: one, the action currents of the nerve itself, and the other, the electrical response of the cochlea.[3]

Hearing is the response of a living animal to physical sound. This response is not purely a physical one but a phenomenon resulting from the ability of the animal to interpret and respond to the stimulation of a sense organ specially adapted for the detection of sound waves. Thus hearing consists, functionally, of two parts: (1) The ability of the sense organ to respond to sound that reaches it. (2) The ability of the brain to interpret the messages or nerve impulses sent to it by the sense organ.

To the sense organ the manner in which sound reaches it, by air, direct transmission through the body or bone vibration, is of no importance. When sound reaches this special organ and stimulates it, then the organ responds by changing this physical stimulation into a biological phenomenon known as the nerve impulse or action current. These nerve impulses travel by a special nerve to the brain. The brain interprets the nerve impulse, resulting in the sensation of hearing. This sense organ for hearing is not the same in all living creatures. There are a great many varieties of hearing organs, but the one we are chiefly concerned with is the organ of hearing for the mammalian group of animals to which man belongs. All mammals have the same sense organ for hearing, and it is called the organ of Corti, after Marchese Corti, who described it in 1851.

By the study of the organ of Corti in animals which have acquired or inherited deafness, one can draw conclusions as to what may happen in the human organ of Corti. It is the results of experimental work and studies on the organ of Corti that I would like to present and point out their application to the causes of human deafness.

Deafness is of two types: (1) Inability of sound to reach the sense organ, which is called conduction, or middle ear deafness. (2) Inability of the sense organ or brain to respond or interpret sound reaching it, which is called perception, nerve, or variable deafness. (I shall use the term nerve deafness to avoid confusion.)

Middle ear or conduction deafness is fundamentally the prevention of air-borne sound waves from reaching the organ of Corti. This results in a loss of hearing for air-borne sounds, but does

---

* Presented at the Meeting of the Acoustical Society of America, Iowa City, Iowa, November 4, 1939.

[1] E. G. Wever and C. W. Bray, "Auditory Nerve Impulses," Science **71**, 215 (1930).

[2] E. G. Wever and C. W. Bray, "The Nature of Acoustic Response: The Relation Between Sound Frequency and Frequency of Impulses in the Auditory Nerve," J. Exp. Psych. **13**, 373–387 (1930).

[3] H. Davis, A. J. Derbyshire, M. H. Lurie and L. J. Saul, "The Electric Response of the Cochlea," Am. J. Physiol. **107**, 311–332 (1934).

not mean that the person is actually deaf. The organ of Corti will respond when the sound reaches it for in this type of deafness the organ of Corti is normal and can respond in a normal manner.

Retraction of the drum as a result of a blocked Eustachian tube is one of the most common causes of conduction deafness. The resulting increase of impedance for the middle ear mechanism causes a loss chiefly of the low tones and in a number of cases also a loss for high tones. This cause of middle ear or conduction deafness may be present in animals but we have no way of detecting it by microscopic examination after the death of the animal. The type of middle ear deafness of which we can be more sure, is caused by infection. Infection fills the middle ear cavity thus blocking the transmission of sounds. Animal experiments showed that the chief cause of the conduction type of deafness is infection causing adhesions or scar tissue to fix the drum or the stapes so that they cannot function. We have found a number of animals with this type of deafness.[4–5] (Fig. 1).

Middle ear or conduction deafness does not present a serious problem to the physicist interested in sound for it is a physical problem that is overcome by any apparatus that can bring sound to the organ of Corti. It is these cases that give the best results with any apparatus used. They are the ones that always hear well on the telephone. It is in conduction deaf cases that we get our best results. If the deaf person has both conduction and nerve (perception) deafness, it is chiefly the improvement obtained for the conduction deafness that gives the patient relief.

FIG. 1. Middle ear of cat. Showing fibrous and inflammatory tissue filling space between drum and stapes causing a deafness of the conduction type.

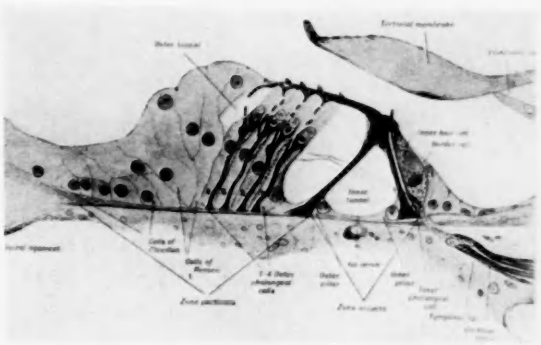

FIG. 2. Diagram of human organ of Corti (after Held) from Maximow and Bloom, 1930 edition, W. B. Saunders.

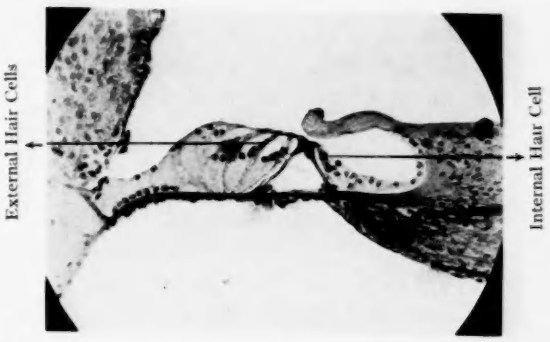

FIG. 3. Normal organ of Corti of dog. Note relationship of external and internal hair cells. External hair cells rest on basilar membrane. Internal hair cell rests on bone.

The nerve (perception) deaf person presents difficulties that seem impossible to overcome. The apparatus developed at the present time have not, as yet, overcome deafness of this type. From experimental evidence I wish to present the possible reasons for failure with this problem.

Nerve (perception) deafness in animals as well as in humans can be acquired or inherited. Acquired deafness in animals can be produced in the same manner in which it occurs in the human, by trauma and poisoning either by drugs or disease.

Before taking up the results from experimentation on the organ of Corti, a description of some of the histological structures of this organ may

[4] M. H. Lurie, H. Davis and A. J. Derbyshire, "The Electrical Activity of the Cochlea in Certain Pathological Conditions," Annals of Otology, Rhinology and Laryngology **43**, 321–344 (1934).

[5] M. H. Lurie, "Animal Experimentation on Hearing: Its Clues to the Prevention of Deafness," Trans. Acad. Ophthal. and Otolaryng. (1935).

be of help. The great part of the organ of Corti rests on the basilar membrane. The organ of Corti, when seen in cross sections, has an internal hair cell, internal pillar cell, external pillar cell and 3–4 external hair cells. The other cells of the organ of Corti are supporting cells. The cells supporting the external hair cells are called Deiters cells and outside of these cells are supporting cells called Hensen and Claudius cells. The supporting cells of the internal hair cell are the internal sulcus cells. One very important fact is that the internal hair cell and internal pillar cells rest on bone and not on the basilar membrane. The external pillar cells and external hair cells rest on the basilar membrane. From this difference in the supporting structure of the hair cells there is an indication of possible difference of function of the internal and external hair cells (Figs. 2 and 3).

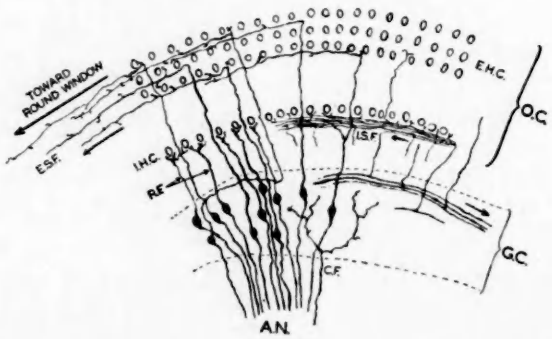

FIG. 4. Diagram of the innervation of the organ of Corti (after Lorente de No). O.C. organ of Corti; E.H.C., external hair cells; E.S.F., external spiral fibers, each innervating many external hair cells; I.S.F., internal spiral fibers, unknown function; I.H.C., internal hair cell; R.F., radial fibers innervating internal hair cells; G.C., ganglion of Corti. Arrows show direction of the fibers away from their cell bodies. C.F., centrifugal fibers of unknown function; A.N., auditory nerve.

The nerve supply of the external and internal hair cells is also different. The internal hair cells are supplied singly or at the most two by a single nerve fiber. A great number of the external hair cells, however, are supplied by a single nerve fiber. This innervation of the external hair cells has been traced for a quarter of a turn in the cochlea by Lorente de No[6] (Fig. 4). Thus the nerve supply also indicates a possible difference of function for these two types of sensory cells.

[6] R. Lorente de No, "Anatomy of the Eighth Nerve," Laryngoscope **43**, No. 1 (1933).

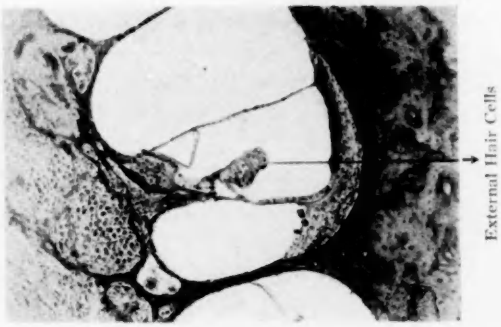

FIG. 5. Organ of Corti of guinea pig exposed to a sound of 400 at intensity of 120 db. Note the lifting off of the external hair cells without rupture of the basilar membrane.

If this is so, can one show the difference experimentally? Before discussing this, the question of vibration of the basilar membrane must be settled as it has an important bearing on this subject.

In the experimental work on the placement of pitch on the basilar membrane,[7] the organ of Corti was first thrown off the basilar membrane by the use of sudden intense sounds (120 db) the pitch being 400 c.p.s. The organ of Corti was thrown off without rupturing the basilar membrane. A number of these experiments gave the same result. Later repetition of this experiment was done by gradually increasing the intensity to 120 db until there was no recovery from the loss of response. Examination of the cochlea showed that the external hair cells of the organ of Corti had been lifted off the basilar membrane. These experiments and others showed that the tones of middle range were localized in the middle two-thirds of the basilar membrane, high tones at the basal end and low tones at the helicotrema or top of the cochlea (Fig. 5).

With the proof of the basilar membrane vibrating and throwing the external hair cells off, experiments were carried out to make animals (guinea pigs) deaf by exposing them to loud sounds for long periods of time.[8] In these experiments a sound of 2500 c.p.s. intensity of 100 db was used for about a month. Animals (guinea

[7] S. S. Stevens, H. Davis and M. H. Lurie, "The Localization of Pitch Perception on the Basilar Membrane," J. Gen. Psych. **13**, 297–315 (1935).

[8] H. Davis, A. J. Derbyshire, E. H. Kemp, M. H. Lurie and M. Upton, "Functional and Histological Changes in the Cochlea of the Guinea Pig Resulting from Prolonged Stimulation," J. Gen. Psych. **12**, 251–278 (1935).

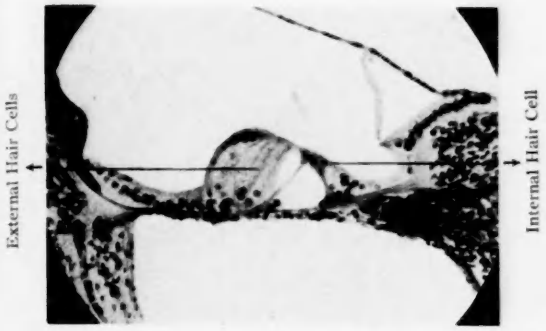

FIG. 6. Organ of Corti of guinea pig. Exposed to a tone of 2500 intensity 100 db for one month. Complete degeneration of external hair cells. Note internal hair cell is normal.

pigs) subjected to this exposure were then tested for the electrical response of the organ of Corti.

The microscopic study of these ears showed changes in the organ of Corti that could be correlated with the loss found by the electrical response. The external hair cells had degenerated completely where the tone loss was greatest, and as this loss approached normal, the organ of Corti showed more normal external hair cells present until a normal organ of Corti was present with a normal response. The internal hair cells were found to be normal (Fig. 6). Here were experiments showing that the exposure of the organ of Corti to loud noises caused a degeneration of external hair cells with loss of sensory response similar to the loss of hearing found among men working in factories, where loud noises are present. One must recognize that these noises are distinct factors in causing loss of hearing. This type of deafness is likely to occur in riveters, boilermakers, blacksmiths, aviators, and workers in other noisy industries. This has been known for a great many years, and was first called blacksmith's and later, boilermaker's deafness.

I would like to point out that the organ of Corti of these animals did respond to sounds but only at higher intensities than normal. This is also true in human beings. What sensory cells could still respond? From our evidence we believe it to be the internal hair cells, for they rest on bone and are less subject to the vibration of the basilar membrane. Further evidence was obtained from animals that were made deaf by the feeding of quinine. They showed degeneration of the external hair cells. These animals also were tested for deafness and electrical response and found deaf. The internal hair cells were normal[5, 9] (Fig. 7).

A few animals were found accidentally in the course of experiments that gave loss of response from 30–40 db throughout the range. These animals also, when examined histologically, were found to have degeneration of the external hair cells. Internal hair cells were found normal. It is these animals that have given direct evidence for the function of the internal hair cells. The cause of the loss of external hair cells is still not known, probably some systemic disease (Fig. 8).

Thus from an experimental viewpoint the chief cause of nerve (perception) deafness is a degeneration of hair cells of the organ of Corti. The external hair cells going first, causes a partial loss of hearing, and when both external and internal hair cells degenerate, a complete loss of hearing results. With the loss of external hair cells there is a loss of sensitivity of about 20–30 db. It was also found that internal hair cells begin to respond to intensities between 30–40 db above threshold.

It is of interest that this conception of nerve (perception) deafness being due to a degeneration of external hair cells is confirmed by the work of Steinberg and Gardner[10] at the Bell Laboratories. They showed that, in the person with

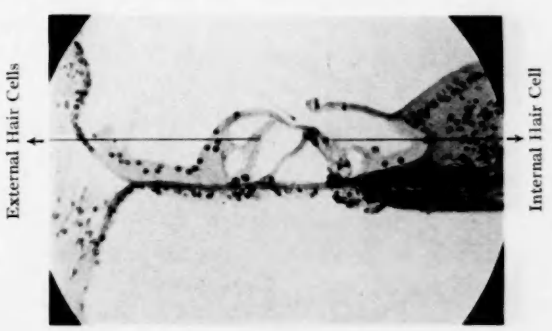

FIG. 7. Organ of Corti of guinea pig made deaf by feeding with quinine for a period of two months. Note complete degeneration of the external hair cells and beginning degeneration of internal hair cell.

[9] M. H. Lurie, "What is Perception Deafness from a Physiological and Histological Basis?" Annals of Otology, Rhinology and Laryngology **48**, No. 1, p. 3, March (1939).

[10] J. C. Steinberg and M. B. Gardner, "The Dependence of Hearing Impairment on Sound Intensity," J. Acous. Soc. Am. **8**, 176–180 (1937).

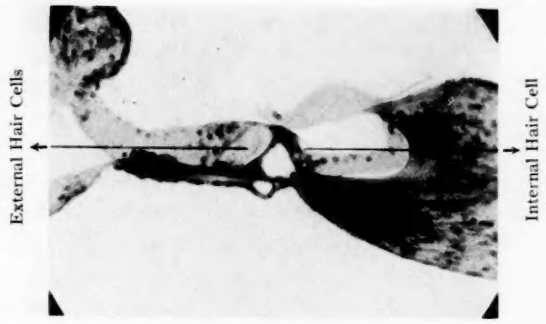

FIG. 8. Organ of Corti of cat showing degeneration of external hair cells of unknown cause. Electrical response gave a raised threshold from 30–40 db throughout the range tested.

nerve (perception) deafness, testing done at intensities of 70 db above threshold for loudness discrimination, the discrimination was the same as for the normal person; but the deafness due to the conduction apparatus still showed the original loss at all intensities (Fig. 9).

From our experimental work we also came to the conclusion that the function of the external hair cells is not so much pitch discrimination as to detect weak sounds. The internal hair cells being more for accurate pitch discrimination and loudness. This conception has been based on the work of Shower and Biddulph[11] who have reported on pitch discrimination in the human. They found that a person could discriminate about 500 different pitches when the intensity used was below 20 db, while at 60 db the pitch discrimination was about 1500.

In all this work done on animals attention must be called to the fact that it is only the electrical response of the organ of Corti that was studied, but it is an accurate measure of what happens in the sense organ. What the animal actually hears can only be measured by the conditioned reflex, that is a response by the animal to the sound heard.

To summarize these experiments: (1) Basilar membrane does vibrate. (2) Animals can be made deaf by acoustic trauma, disease or toxic poisoning. (3) In all these deafened animals the type of lesion found in the organ of Corti was the same.

(a) Degeneration of the external hair cells resulting in a loss of response similar to human nerve (perception) deafness. When both external and internal hair cells degenerate in a specific area there is a tonal gap which can be bridged by increasing the intensity of the sound.

(b) The internal hair cells can and do function in the presence of degenerated external hair cells.

To build apparatus to improve the hearing of the nerve (perception) deaf person, these factors must be kept in mind: that the deafness is not due chiefly to inability of sound to reach the organ of Corti but to the fact that the sense organ or the nerve itself is involved. It is for this reason that those afflicted with this type of deafness find very little comfort from hearing devices. They are always complaining that the apparatus makes all sounds and noises too loud. With loss of the external hair cells there is a loss of the ability to pick up weak sounds, and, therefore, lack of response to those weak high frequency sounds made by the letters, such as, s—t—f—c, etc. To enable these deaf people to hear, the apparatus must be built up to a point where the remaining hair cells are stimulated. When these remaining hair cells are stimulated, the sounds heard are loud and disagreeable for there is but a small range between threshold and extra loudness as shown by Steinberg's curve. The extra loudness needed for hearing by the nerve (perception) deaf person causes irritation and sensation of pain.

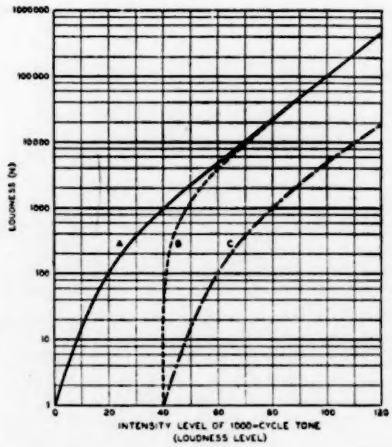

FIG. 9. Loudness *versus* intensity level of a 1000-cycle tone for (*A*) normal, (*B*) nerve deafened, and (*C*) conductive deafened observers (Steinberg and Gardner).

[11] E. G. Shower and R. Biddulph, "Differential Pitch Sensitivity of the Ear," J. Acous. Soc. Am. **3**, 275–287 (1931).

The other type of nerve deafness is inherited deafness. This, too, has been found in animals. I exclude from this type otosclerosis as we have not found true otosclerosis in animals, up to the present time, nor have we been able to reproduce ankylosis of the stapes except by direct trauma or infection. A number of animals with inherited deafness were studied, cats, dogs, mice and guinea pigs.

The cats studied were of the albino type and were found deaf on the side with the blue eye. The electrical response of the organ of Corti was negative and sections showed complete absence of the organ of Corti.

The dogs studied were of special Collie type bred for its silvery blue color, and a number were found to be born deaf. Testing of these animals showed a loss of hearing in one or both ears. Histological examination showed absence of normal organ of Corti but normal nerve going to the basilar membrane.

Mice that inherit deafness also inherit a tendency to move in circles, and are familiar to all of you as the Japanese waltzing or dancing mice, which can be bought at any pet shop. The study of the ears in these mice show changes that are similar to the ears of the waltzing guinea pig.

The waltzing guinea pig has been of great interest as it has given clues as to what may occur in the human ear.[12] The circling of the animals is inherited and is not due to any pathological changes in the vestibular apparatus, which has to do with equilibrium. The cause of the circling we believe is in the midbrain of the animal, and we have succeeded in making normal guinea pigs circle by producing a lesion in a particular area of the midbrain.[13]

The deafness of these animals is inherited and follows the Mendelian laws for recessive characteristics. This law of recessive characteristics is that the offspring of a mating will always produce animals which have the dominate characteristics of the two animals. But when the hybrids are mated then one of their offspring will again reproduce characteristics that have been hidden.

FIG. 10. Organ of Corti, waltzing guinea pig. Upper turn of cochlea. Showing degeneration of external hair cells with a normal appearing internal hair cell. This cochlea had a normal organ of Corti at the helicotrema and a complete absence of organ of Corti at the base.

For example, a normal guinea pig mated with a waltzing guinea pig gives hybrids that appear and act normally. They do not circle nor are they deaf. But if we take two of these hybrids and mate them, the litter from this mating will, as a general average, produce one normal animal, two animals which are a mixture of the strains but act normally, and one animal that will waltz and be deaf.

The studies of these waltzing guinea pigs show that the animal develops a normal organ of Corti during intra-uterine life but when it is born or near birth the organ of Corti begins to degenerate. The degeneration that occurs is similar to that described in animals made deaf experimentally. The external hair cells go first, then the internal hair cells and finally complete absorption of the organ of Corti, leaving nothing but a single layer of cells on the basilar membrane. The nerve and ganglia cells do not degenerate until later. These animals when tested give no electrical response except when there is some normal or almost normal organ of Corti present at the helicotrema or top of the cochlea (Fig. 10).

The fact that this degeneration starts after the normal development of the organ of Corti is of great importance for it points to a way human beings become deaf, that is by a direct inheritance factor. Deafness of this type can start at any age and gradually become severe, in all probability going through the stages found in these animals. For these people the outcome is hopeless as we can do nothing with this condition.

[12] M. H. Lurie, "Studies of the Waltzing Guinea Pig," Laryngoscope, St. Louis, July, 1939.

[13] M. H. Lurie and E. W. Dempsey, "Experimentally Induced Circling (Waltzing) in the Guinea Pig," Laryngoscope, St. Louis, July, 1939.

Conclusions

(1) Conduction deafness is a deafness due to mechanical interference with the transmission of sound to the organ of Corti and can be overcome by hearing apparatus.

(2) Nerve (perception) deafness is due to the degeneration of the sensory hair cells in the organ of Corti. (a) A degeneration of external hair cells causes loss of hearing up to about 30 db. (b) The internal hair cells are the chief functioning sensory cells in the nerve (perception) deaf person and they respond only to sounds of greater intensity than the external hair cells. (c) With loss of both internal and external hair cells there is complete deafness for the region of the basilar membrane involved.

Apparatus built for the nerve (perception) deaf person must take into consideration these facts. To increase the intensity of all sounds does not improve the hearing of the nerve (perception) deaf person.

Thus, in this rapid survey of our conception of deafness, one can readily see that no cure or hope for cure is presented. The only means we have at present of stopping these conditions is by prevention. Remove or prevent the causes and the hearing can take care of itself. For the acoustical engineers it may be more important from the human standpoint to develop more noise preventing apparatus rather than more sound apparatus to irritate and destroy the delicate hair cells of our already over-stimulated organ of Corti.

The Locus of Distortion in the Ear[1]

ERNEST GLEN WEVER, CHARLES W. BRAY AND MERLE LAWRENCE
Princeton University, Princeton, New Jersey

(Received February 10, 1940)

IT is well known that sinusoidal sound waves applied to the ear are heard as simple tones when they are weak, and as complex tones when they are strong. To a limited extent, a person of musical training can analyze complex sounds thus stimulated, and can recognize the early members of a harmonic series. Such analysis is facilitated, and quantitative measurements are made possible, by the "exploring tone" method, in which an additional tone is adjusted in frequency to give beats with each component in turn.

Since the time that Helmholtz first discussed the problem,[2] the distortion that is subjectively appreciated under the conditions mentioned has generally been regarded as arising peripherally, in the mechanism of the ear. Studies of combination tones, masking, and modulation effects have given presumptive evidence for this view. Final proof has recently been afforded by experiments on animals which made use of the electrical effects generated in the cochlea.[3]

In experiments reported from this laboratory,[4] the ear of the cat was stimulated with a pure tone of 1000~, and was found to yield electrical responses which progressively increased in complexity as the stimulus intensity was raised. At high levels of stimulation, analysis revealed a complete series of partials, extending in some instances as far as the 16th. For great intensities the total overtone content exceeded 50 percent. The general picture of distortion as displayed in such experiments is consistent with the results of psychological studies of human beings.

The above evidence shows that distortion occurs in the peripheral acoustic mechanism, but it does not reveal the particular part of this mechanism that is responsible. Helmholtz considered the drum and ossicles of the middle ear as the most likely seat of distortion. He pointed out that the drum, like all stretched membranes, should operate in nonlinear fashion when its displacements become large, and that the ossicles might be presumed to have a small amount of looseness at their articulations. Most writers since Helmholtz have subscribed to his view.

Some considerations, however, tell against the theory that distortion has its seat in the middle ear. Several investigators have reported that combination tones can be heard by persons who have lost the drum and the two larger ossicles. Békésy[5] was unable by direct measurements to detect overtones or combination tones reflected from the drum. The same investigator could perceive no significant changes in overtones when a steady pressure was exerted on the drum, though difference tones were considerably altered; and he concluded that the middle ear does not produce overtones, though it may enter into the production of combination tones. Results obtained by Stowell and Deming[6] from experiments on modulated tones led them to postulate a rectifier action of the inner ear. In our own experiments on distortion in the ear of the cat it was observed that considerable alterations of the mechanical condition of the middle ear apparatus had no significant effect upon the distortion pattern. Denervation of the tensor tympani muscle, cutting its tendon, eliciting its reflex contractions, and applying artificial tension to its tendon are measures all of which must alter

[1] From the Princeton Psychological Laboratory. This investigation was aided by grants from the National Research Council and the Rockefeller Foundation.

[2] H. L. F. Helmholtz, *On the Sensations of Tone*, translated by Ellis (2nd Eng. ed., 1885), pp. 158–159.

[3] S. S. Stevens and E. B. Newman, "On the Nature of Aural Harmonics," Proc. Nat. Acad. Sci., Washington **22**, 668–672 (1936); E. B. Newman, S. S. Stevens, and H. Davis, "Factors in the Production of Aural Harmonics and Combination Tones," J. Acous. Soc. Am. **9**, 107–118 (1937); E. G. Wever and C. W. Bray, "Distortion in the Ear as Shown by the Electrical Responses of the Cochlea," J. Acous. Soc. Am. **9**, 227–233 (1938); A. F. Rawdon-Smith and J. E. Hawkins, "The Electrical Activity of a Denervated Ear," Proc. Roy. Soc. Med. **32**, 496–507 (1939).

[4] E. G. Wever and C. W. Bray, reference 3.

[5] G. v. Békésy, "Über die nichtlinearen Verzerrungen des Ohres," Ann. d. Physik **20**, 809–827 (1934); "Physikalische Probleme der Hörphysiologie," Elek. Nachr. Techn. **12**, 71–83 (1935).

[6] E. Z. Stowell and A. F. Deming, "Aural Rectification," J. Acous. Soc. Am. **6**, 70–79 (1934).

greatly the operating conditions of the mechanism. None of these, however, was effective in modifying the distortion pattern in any significant way. The last two procedures reduced the general level of response by the introduction of damping, and altered the distortion in proportion to this reduction; but when the stimulus intensity was raised to compensate for the change in response level, the distortion was the same as before. We concluded that the most likely seat of distortion is the inner ear.[7] The experiments to be reported here form a more direct attack upon the problem.

Method and Procedure

To ascertain the relative roles of the middle and inner ear mechanisms in the process of distortion it is necessary first to eliminate the middle ear by surgical means and then to obtain the distortion pattern of the inner ear alone. A comparison of the results so obtained with those of the intact ear indicates the place of origin of distortion.

We have not found it feasible to separate middle and inner ears at the anatomical division point, which is the membrane of the oval window, for the fragile character of this membrane precludes the removal of the foot-plate of the stapes from it. Our separation, therefore, has been made at the joint between incus and stapes. We thus are comparing the action of the normal ear with that of one deprived of drum, malleus, and incus, together with their ligaments and the tensor tympani muscle. In one experiment we eliminated also the stapedius muscle by cutting its tendon, and thus put out of action the whole middle ear except the stapes.

The experiments were performed upon guinea pigs under urethane anesthesia. The middle ear was opened by the lateral route, and the incudostapedial joint was severed. Thereafter the incus was displaced, and the drum and malleus were removed. Then, after slight enlargement of the meatus, the point of a steel needle which was attached to a crystal vibrator was placed in firm contact with the head of the stapes. The animal's head was securely fixed by a clamp attached to the skull.

The vibrator consisted of a number of Rochelle salt sections 0.05″ thick by $\frac{5}{8}$″ wide by 3″ long, cemented together with interleafing foil electrodes to form a crystal of the expanding type $\frac{5}{8}$″ by $\frac{5}{8}$″ by 3″. It was mounted on a heavy lead reaction block and thoroughly shielded. When in use it was held and moved by a special manipulating apparatus. At times a vibration microphone, also a Rochelle salt crystal, but with the inertia type of mounting, was attached to the Bakelite rod that connected the driving crystal to the needle. The vibration microphone was used as a check upon the linearity of stimulation.

Measurements showed the harmonic content of the stimulus to be negligible for low and moderate input currents, and, as expected, to increase rather sharply above a particular level. For the results to be reported here the input level was below the overload point as thus determined. The amplitudes of stimulation available were not sufficient to give a maximum of response for all ears, though they did so for some. They were enough, however, to give a sufficiently extensive picture of the distortion pattern for present purposes.

The technique of applying the needle of the vibrator to the stapes is somewhat delicate. It is necessary to orient the vibrator so that it operates in a normal manner, with its axis in line with that of the stapes. The stapes must maintain its proper position, and must not make contact with the wall of the cochlea, which is nearby. A firm pressure on the head of the stapes is required, to secure the necessary negative movement of the system. It was found that, once a suitable pressure was obtained, slight increase had little or no effect upon the responses until a critical point was reached, at which injury to the inner ear evidently resulted. Little trouble was encountered in this part of the procedure; experience soon showed the appropriate degree of pressure, and the manipulation was aided by watching the responses and increasing the pressure until they were steady and maximal.

The electrical responses of the cochlea were recorded in the usual manner,[8] by an electrode

[7] E. G. Wever and C. W. Bray, reference 3; see also "The Tensor Tympani Muscle and its Relation to Sound Conduction," Ann. Otol. **46**, 947–961 (1937).

[8] E. G. Wever and C. W. Bray, "The Nature of Acoustic Response: the Relation between Sound Intensity and the Magnitude of Responses in the Cochlea," J. Exp. Psych. **19**, 129–143 (1936).

on the membrane of the round window. These responses were amplified, and studied with a wave analyzer of the heterodyne type (General Radio Type 736–A). Analyses were made for stimulating frequencies of 300~, 1000~, and 3000~, at intensities which extended by 5-db steps from the lowest that gave measurable effects to the highest available without overloading the vibrator, a range of about 80 db. The analysis was carried only as far as the 5th partial in all but one instance, where it was extended to the 10th; the more limited series gives a sufficiently close representation of the distortion pattern for the purposes of the experiment.

Analyses were made also with stimulation in the usual manner by air-conducted sounds from a loudspeaker, with the drum and ossicular chain intact. These results serve as "controls," to represent the distortion of the ear as a whole, and to form a basis for comparison with the pattern obtained from stimulation of the stapes.

Results

Results were obtained on nine ears. Of these, three were normal, and were stimulated with aerial sounds. The remaining six had drum and outer ossicles destroyed, as described, and were stimulated at the stapes. Since, in several instances, analyses were made with more than one stimulating frequency, there were in all seven analyses with aerial sounds and twelve with stimulation at the stapes. Sample results are presented in the accompanying figures.

In these figures the magnitude of cochlear response is shown on the ordinate, in microvolts at the round window membrane, calculated on the basis of calibrations of the amplifying and recording apparatus. Intensity of the stimulating tones is represented on the abscissa, in decibels above and below a reference point which is approximate and not directly determined, since no calibration was available for the vibrator. This reference point was located through a comparison of the level of responses obtained here and in previous studies on guinea pigs in which the stimulus intensities were known. Zero db on the curves represents the intensity which yields the same response ordinarily obtained with a sound of 1 dyne per sq. cm. The results with aerial conduction, which are used for comparison, are represented in the same manner. The individual differences among the animals are sufficiently small to justify this procedure.[8]

Figure 1 represents an analysis of responses evoked by a 1000~ stimulus conducted aerially to a normal ear. The curve marked "1" is for the fundamental frequency, and the remaining curves are for the next four partials as indicated. The fundamental curve is a power function with an exponent very nearly unity except at its upper end, where bending occurs and the response approaches a maximum. The overtone components also are power functions for a part of the range, and bend over at their upper ends. Their curves, however, have slopes considerably above unity, which means that as stimulus intensity rises the harmonics increase relative to the fundamental. In the ear shown, the 3000~ component has a larger slope than the 2000~ component; when the 3000~ component is first measurable it is much below its neighbor, but as intensity is increased it rises rapidly and crosses the 2000~ curve. In this ear there is not much difference between the 4000~ and 5000~ components. However, in many ears studied the 5000~ component, like the 3000~ component, starts at a low level but has a steeper slope and rises above the partial just preceding. In our earlier experiments on the cat, whose ear yields somewhat greater potentials for study, this difference in the behavior of odd and even partials could frequently be discerned as high as the 8th and 9th.[9] The difference appears to be a common though not an invariable characteristic of the distortion pattern.

Figure 2 presents the results of an analysis with stimulation at 1000~ with the vibrator on the stapes. The arrangement is the same as in the preceding figure. The close similarity of the distortion patterns in the two cases is apparent. Here, as before, the fundamental curve has a slope of approximately unity throughout most of its course, and bends at high intensities. The overtones all have slopes above unity. The 3000~ curve starts below that for 2000~, rises somewhat more rapidly, and crosses it as the 2000~ curve goes through an inflection. The 5000~ curve here also starts below the 4000~ curve,

[9] Cf. Fig. 1, p. 228, J. Acous. Soc. Am. 9, (1938).

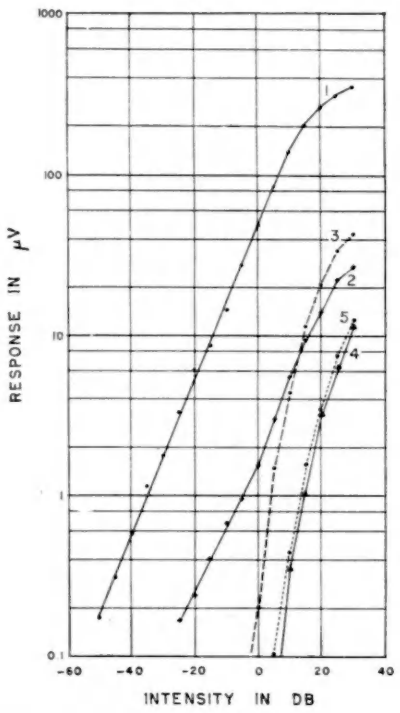

FIG. 1. Analysis of cochlear responses produced in a guinea pig ear by a pure, air-conducted tone of 1000~. Animal No. 276 R.

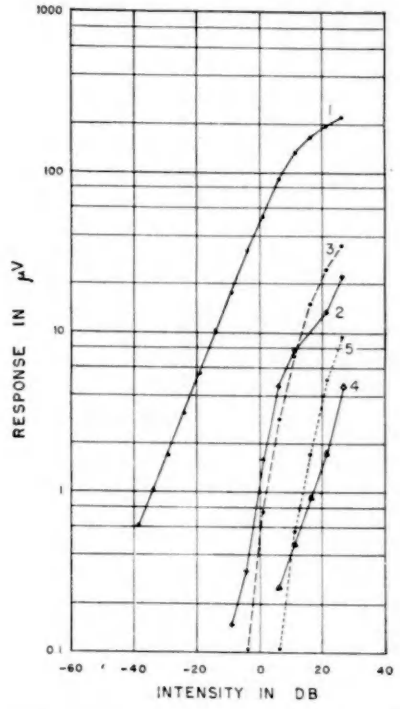

FIG. 2. Analysis of cochlear responses produced in a guinea pig ear by mechanical vibrations applied directly to the stapes. Distortion in the stimulus itself was negligible; the largest component was 2000~, which at maximum intensity was 0.26 percent of the fundamental. Animal No. 275 R.

but soon crosses it. The most significant feature is that the magnitude of distortion, as shown by the general positions of the overtone curves relative to the fundamental, is closely similar for the two forms of stimulation.

The next two figures afford a similar comparison of distortion patterns for stimuli of 3000~. Fig. 3 shows the results of an analysis, in an intact ear, with aerial stimulation. As before, the components are power functions for the lower part of their courses and depart from this form at their upper ends. Through most of its extent the fundamental curve has a slope of approximately unity, while the overtones have steeper slopes. Here the 3rd partial does not cross and rise above the 2nd, though it does rise high enough to coincide with the other in mid-course. A similar relation holds for the 4th and 5th components. Somewhat more bending is evident at the upper end of these curves than in the preceding figures; the difference is due in part to the greater intensity available for stimulation.

Figure 4 presents results for stimulation at 3000~ with the crystal vibrator in contact with the stapes. Here two curves are given for the fundamental, representing results obtained at different times. Through most of the range, the only difference is a slight drop of sensitivity, amounting to less than 1 db, for the second set of measurements. At the higher stimulus intensities, however, the curves differ markedly in form: one bends over and sharply downward, while the other, which represents the later measurements, bends only moderately.

The overtone curves show characteristics most of which have already been described. Inflections are present in both the 2nd and 4th partials. A new detail is that the rise after the inflection is so great that a second crossing with the neighboring odd component occurs. This character has also been found repeatedly, for both types of stimulation.

A significant point in regard to the results shown in Fig. 4 is that in this animal all activity of the stapedius muscle was precluded by cutting its tendon. This latter operation was not done in

the other experiments because it removes all suspension of the stapes except at the oval window, and renders the manipulations of the stimulator particularly difficult. Five analyses, at three different frequencies, were made on this animal, and their results duplicate in all essential respects those obtained from animals in which the stapedius tendon was intact. The experiment therefore is crucial: the stapedius muscle is not responsible for the distortion pattern as here exhibited.

Discussion

The results show no essential differences between the distortion pattern for the intact ear, stimulated by aerial sounds, and that for the ear deprived of all middle ear structures except the stapes and stimulated by mechanical vibrations. The same components are present, in about the same degree. Variations are observed from one ear to another, and to some extent within measurements on a given ear, but these are no more characteristic of one experimental condition than the other. Hence we conclude that the general pattern of distortion that is portrayed by the electrical responses of the cochlea has its origin beyond the stapes, and not in the middle ear.

In attributing the distortion pattern to the inner ear we are assuming that in our tests after middle ear destruction the distortion observed was not an artifact of stimulation. It is evident that failure of the stapes to follow the needle point would give extraneous effects. It is unlikely, however, that a distortion pattern so introduced would resemble that manifested by the intact ear for aerial sounds. Moreover, the distortion pattern appears highly resistant to mechanical changes of the system. We have already referred to our earlier observations in which manipulations of the tensor tympani muscle were without significant effect upon distortion. The present experiments give further evidence. Slight variations of contact pressure between vibrator and stapes had no noticeable result, provided that the pressure was adequate. Insufficient pressure was indicated by the fact that large variations of response occurred in synchronism with respiration. However, despite the variations under these

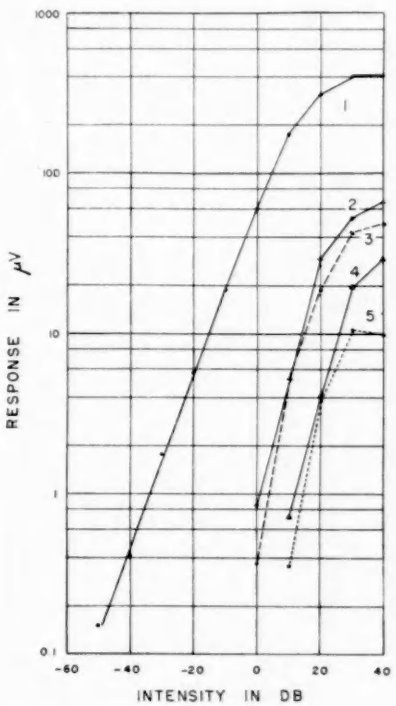

Fig. 3. Analysis of cochlear responses to an air-conducted tone of 3000~. Animal No. 263 R.

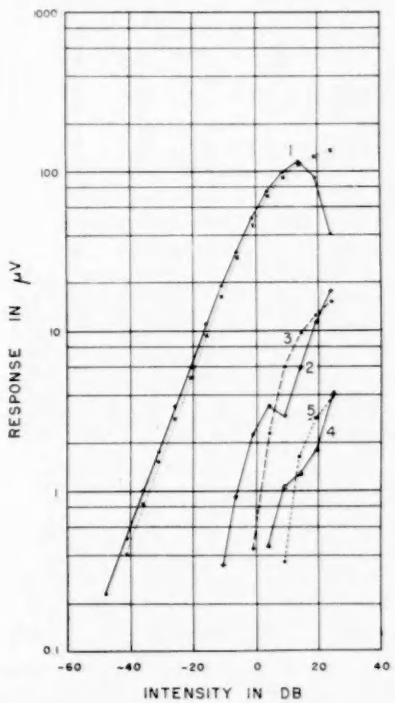

Fig. 4. Analysis of cochlear responses with direct stimulation at the stapes, with a frequency of 3000~. The stapedius tendon was cut. Animal No. 270 R.

conditions, readings taken during quiet moments between breaths gave a distortion pattern of the usual kind. When a suitable pressure was employed, the variations with breathing ceased.

Our conclusion that the distortion pattern here represented has its origin beyond the stapes need not imply that the middle ear does not distort at all, but rather that its effects are negligible in relation to the inner ear effects. It is possible that for levels of stimulation beyond those here employed additional distortion of middle ear origin might be discernible. It may be significant, however, that for strong intensities the inner ear reaches a maximum of effectiveness, and thus provides an upper boundary to the requirements made on the more peripheral conductive system.

Further consideration is needed of the matter of the variability of responses. Variations of the order of a few db are common for fundamental and overtones alike at high intensities of stimulation. The fundamental curve sometimes bends only moderately, and at other times passes rapidly through a maximum as the stimulus intensity reaches high values. Variations are even more common for the overtones. The even harmonics, in particular, are subject to many irregularities, and often exhibit a flexure in mid-course. It is evident that great instability characterizes the action of the ear when it is severely overloaded. This instability is not caused by spasmodic contractions of the middle ear muscles, for it remains after these parts have been lost; it must lie within the inner ear itself.

The results thus do not support certain conclusions reached by others who have studied distortion in the cochlear response. Stevens and Newman,[3] and later Newman, Stevens, and Davis,[3] supposed that tension of the middle ear muscles altered the distortion pattern by reducing even harmonics and leaving odd harmonics unchanged. In a guinea pig they increased the depth of anesthesia and found a decrease of second harmonic. In a cat they cut the tensor tympani tendon and observed an increase of second harmonic; in both instances the third harmonic was unaffected. It should of course be noted that this evidence is itself not wholly consistent, inasmuch as muscular relaxation during deep anesthesia should operate in the same direction as cutting the tendon, and not the reverse. We consider the variations which they observed to be those commonly found in the responses of even harmonics, and to bear only an incidental relation to the preceding experimental measures.

More recently, Rawdon-Smith and Hawkins[3] made observations on a cat's ear in which an operation had been made to damage the 8th nerve, and found a harmonic pattern which they regarded as significantly different from that observed in the same laboratory by Newman, Stevens, and Davis. The differences were largest (about 5 db) for the second harmonic: the denervated ear gave less of this component at low intensities, and more at high intensities, than the normal ear studied previously. It was suggested that the difference arose from the inactivity of the tympanic muscles in the denervated ear. However, an explanation in terms of middle ear muscles is not demanded by the facts. As already mentioned, variability in the second harmonic is common; and here we have different animals studied by different experimenters. Moreover, as Rawdon-Smith and Hawkins showed, the electrical output of the cochlea was somewhat altered in the denervated ear due to the elimination of a nervous component in the response.

It must be made clear that contractions of the tympanic muscles do reduce the transmission of overtones along with the fundamental[10] Also, for a given stimulus it is easy to observe that the percentage reduction of the overtones is greater than that for the fundamental; but this relation arises simply from the nature of the distortion pattern itself and its change with the level of stimulation. As has been shown, an increase of stimulus intensity raises the overtone content relative to the fundamental, and it naturally follows that any measure that reduces transmission will retrace this path. There is as yet no evidence that the middle ear muscles affect the harmonic pattern except in direct relation to the change of transmission.

The results indicate distortion as arising beyond the stapes, but they do not further specify

[10] First pointed out by C. S. Hallpike, "On the Function of the Tympanic Muscles," J. Laryngol. and Otol. **50**, 362–369 (1935).

its origin. It could arise mechanically at the oval window, in the fluid pathways of transmission or in the movements of membranes; or it could appear in the process through which mechanical is converted into electrical energy. We favor the last-mentioned possibility in view of the considerable independence of distortion from mechanical changes. This independence, furthermore, points to the hair cells of the organ of Corti as the most probable site of the transformation of mechanical into electrical energy.

Summary

An investigation was made of the locus of distortion in the ear by the surgical separation of middle and inner ear structures at the stapes. Stimulation of the isolated inner ear by mechanical vibrations gave a distortion pattern of the same type as that obtained from the intact ear by aerial stimulation. It is concluded that distortion in the middle ear is insignificant, and that it has its seat of origin beyond the stapes, perhaps in the hair cells of the organ of Corti.

The Genesis of Absolute Pitch

A. BACHEM
College of Medicine, University of Illinois, Urbana, Illinois
(Received January 25, 1940)

A STUDY[1] of 103 cases of absolute pitch revealed several types of this faculty:

Genuine absolute pitch is based upon immediate recognition of the chroma, i.e., the quality which a tone has in common with its lower and higher octaves (*C*-ness, *D*-ness, etc.). This is the most frequent type of absolute pitch.

Quasi-absolute pitch is based upon an aural or vocal standard, and the interval sense. This type is occasionally found in orchestra players and singers, respectively.

Pseudo-absolute pitch is based upon the estimation of tone height. While most persons can roughly estimate the position on the tonal scale of a tone heard, some people can be trained to make rather accurate statements, at least for a short time after the training period. This faculty has been described by several authors as absolute pitch, although it differs from the latter in many ways.

All these types of absolute pitch occur with rather pronounced variations in acuity, range of the musical scale, influence of timbre, etc.[1] The discussions in this paper will deal chiefly with genuine absolute pitch, as the type to which most authors and musicians refer as "absolute pitch" in a narrower sense.

I

Several authors claim that absolute pitch can be acquired by training: Meyer[2] found that persons can be trained to identify 9 tuning forks (large intervals) correctly in 83 percent of the judgments, but 16 forks (smaller intervals) in 56 percent only. Köhler[3] claims that he acquired absolute pitch in 11 to 14 days of training. His judgment errors show a random distribution up to seven half-tones. Gaugh[4] could improve "absolute pitch" from an average error of 5.5 to one of 4.5 half-tones. Mull[5] taught 6 persons to identify the middle *C* of the piano correctly in 82 percent of the cases. Wedell[6] deduced from his elaborate experiments that unmusical persons can learn to recognize tones $8\frac{1}{3}$ semi-tones apart. The limit of ability reached with more musical persons was an average judgment error of 3 semi-tones.

Obviously all these authors were not dealing with genuine, but with pseudo-absolute pitch. These authors did not possess absolute pitch themselves, nor did they use a person with absolute pitch for comparison.

The outstanding difference between the "absolute pitch" of these authors and genuine absolute pitch lies in the accuracy of judgment. While the average error of judgment in these cases of "pseudo-absolute" pitch amounts to about 5 half-tones (after training), the corresponding values in case of genuine absolute pitch are 1/11 of a half-tone according to Abraham,[7] and even less according to other authors.

To secure comparative results, identical experiments were conducted on a similar number of persons without and with absolute pitch. A first pair of experiments was performed on the piano by offering a tone to a person placed with the

TABLE I.

SUBJECT	HALF-TONE ERRORS, DAILY					AVERAGE
Mr. S.	5	7	5	4	5	5
Dr. G.	7	Not continued				—
V. B.	6	7	6	3	4	5
Mr. M.	6	4	10	8	7	7
Dr. V.	6	Not continued				—
Mr. A.	4	6	6	5	5	5
Dr. C.	9	Not continued				
Average						5+

[1] A. Bachem, "Various Types of Absolute Pitch," J. Acous. Soc. Am. **9**, 146 (1937).
[2] M. Meyer, "Is the Memory of Absolute Pitch Capable of Development by Training?" Psychol. Rev. **6**, 514 (1899).
[3] W. Köhler, "Akustische Untersuchungen III," Zeits. f. Psych. **72**, 159 (1915).
[4] E. Gaugh, "The Effects of Practice on Judgments of Absolute Pitch," Arch. of Psychol. **7**, 1 (1922).
[5] K. Mull, "The Acquisition of Absolute Pitch," Am. J. Psychol. **6**, 469 (1925).
[6] C. H. Wedell, "Nature of Absolute Judgment of Pitch," J. Exp. Psychol. **17**, 485 (1934).
[7] O. Abraham, "Das absolute Tonbewusstsein," Sammelbde d. internat. Musikges. **3**, 1 (1901).

TABLE II.

	ABSOLUTE PITCH			NO ABSOLUTE PITCH		
SUBJECT	DR. B.			MR. A.		MR. S.
OFFERED	*e* 650	*g* 730	*a* 870	*e* 650		*a* 870
JUDGED	±VIBRATIONS PER SECOND			±VIBRATIONS PER SECOND		
After 5 min.	+ 8	0	−12	−70		−30
After 10 min.	− 4	+5	0	+45		−50
After 15 min.	− 3	−3	+ 3	+40		−70
After 20 min.	− 1	−4	− 8	+40		0
After 25 min.	+ 4	+2	+ 5	−24		−20
After 30 min.	+10	+6	+ 3	+25		+55
Av. Err.	5	3.3	5	41		38
SUBJECT	DR. B.			MR. A.	MR. S.	MR. W.
OFFERED	*e* 650	*g* 730	*a* 870	*e* 650	*a* 845	*a* 845
JUDGED						
After ½ day	− 5	+3	− 3		−25	
After 1 day	− 2	−4	+ 6	+56	+25	+ 5
After 1½ days	− 4	0	−10		−50	
After 2 days	0	+2	+ 3	−34	−33	− 50
After 2½ days	+ 5	+5	+10			
After 3 days	− 1	0	+ 7	+26	−10	−160
After 3½ days	−12	−8	+13			
After 4 days	+ 8	+5	−10	+60	−80	−238
After 4½ days	+ 3	+2	0			
After 5 days	+ 2	+9	− 3	+65	+21	
Av. Err.	4.2	3.8	6.5	48	35	113
Total Av. Err.	4.6	3.6	5.8	44	37	113
Error in percent of tone frequency	0.7%	0.5%	0.6%	6.7%	4.3%	13.4%

back toward the piano and then having the person turn toward the piano and identify the key without striking it. The average of 20 judgments was calculated and another 20 tones were offered and identified by striking the key, in order to train the person for the same experiment on the following day. The results are tabulated on Table I for persons without absolute pitch.

In order to compare these results with the accuracy of absolute pitch judgment, several ranges of the piano scale were tuned in eighth tones, and marked correspondingly. Most subjects identified the tones correctly.

A second pair of experiments was designed for more accurate results. A tone was offered several times on a tone variator which was standardized in vibrations per second. Every five minutes a different tone was sounded and altered according to directions of the subject, until the tone was considered as the right one. A similar experiment was conducted with the five-minute intervals replaced by half-day intervals.

Table II shows the results for persons with and without absolute pitch:

In these experiments the average error of absolute pitch determination was about 0.6 percent for observations extending over ½ hour, as well as for those extending over five days. An identical experiment with two nonprofessional and three professional musicians gave similar results:

Nonprofessional	Dr. W. Z.	0.6%	1/9 half-tone
	J. R. T.	1.0%	1/5 half-tone
professional	St. R.	0.35%	1/16! half-tone
	A. N.	0.4 %	1/14 half-tone
	H. K.	0.5 %	1/11 half-tone
	Avg:	0.6 %	1/9 half-tone

For persons without absolute pitch the error amounted to about 8 percent on the tone variator, representing an interval of 1 to 2 half-tones.

On the piano, however, an error of about 5 half-tones was found.

(The noticeable difference between the accuracy of pitch judgment of persons without absolute pitch on the piano and on the tone variator is due to the fact that the change of timber, brightness, intensity with frequency is decidedly stronger on the tone variator than on the piano. Another reason for the better judgment of nonmusical persons on the tone variator is the fact that a single tone was observed on that instrument upon which attention and memory was concentrated, while on the piano different tones were offered. With the one tone method used on the piano a 4-half-tone average error resulted, in contrast to one of 5 half-tones if tones all over the piano scale were offered.)

These results give conclusive evidence of the great difference in accuracy of pitch identification in persons with and without absolute pitch.

It is the opinion of v. Kries[8] that a certain accuracy of tone memory amounting to less than a half-tone is necessary to render absolute pitch determination possible. If such an acuity is lacking, identification is impossible. V. Kries, however, fails to explain the difference of magnitude in the judgment error of those with and without absolute pitch. This difference is due to the fact that a different mechanism is used for the pitch determination by the two different types of persons. Those with absolute pitch possess the fine method of recognizing tones by their chroma. Those without absolute pitch have only the crude tone height estimation at their disposal for identifying the tones.

Beside this different order of magnitude in the accuracy of tone determinations, other differences are apparent between pseudo and genuine absolute pitch:

Most observers of "acquired" absolute pitch mention the slowness of judgment and the tendency to correct previous judgments. In contrast, the statement in genuine absolute pitch is very fast and decided.

The experiments on acquired absolute pitch were conducted on one single tone or on a limited number of tones on one musical instrument. Even under these simplified conditions the resulting scores were poor. These subjects would have been completely at a loss if confronted with octaves, chords, various keys, and different instruments. Several authors found greater accuracy of judgment at both ends of the musical scale, in contrast to that noticed with genuine absolute pitch.

In summary, the statement that absolute pitch can be acquired by training refers to pseudo-absolute pitch, the crude estimation of tone height, but not to genuine absolute pitch.

In quasi-absolute pitch, one has to assume that training plays an important role. Some singing experience is required, before a vocal standard develops; long experience in tuning a violin and hearing the "a_1" offered to the orchestra is necessary for a more or less accurate judgment of "a_1," and some musical experience is needed for the development of an interval sense, i.e., a correct comparison between the standard and the offered tone. But all this has nothing to do with genuine absolute pitch.

The estimates of Cheslock,[9] that two percent of musical students up to 7 years, five percent between 17 and 21 years, 20 percent of very advanced students and teachers possess absolute pitch, if correct, would prove that this faculty is acquired in the course of musical activity. Cheslock's discussion deals with perfect relative pitch, however, rather than with absolute pitch, and the figures quoted are not substantiated by his study nor by those of any other authors.

Several authors have called attention to the fact that great musicians had no absolute pitch, that many musicians have tried systematically to acquire absolute pitch, without results, and that no case is known of the acquisition of (genuine) absolute pitch by training.

II

Many authors have come to the conclusion that heredity plays a role in the occurrence of absolute pitch.

Stumpf[10] thinks that there is "an hereditary factor involved in the ability of absolute pitch discrimination, although much depends upon memory and even experience." V. Kries[8] discusses the question of acquisition of absolute pitch and concludes that "an individual factor is of greater importance than musical experience." Abraham[7] postulates an individual factor involved in the development of absolute pitch. Révész[11] describes the perfect absolute

[8] J. v. Kries, "Über das absolute Gehör," Zeits. f. Psych. **3**, 259 (1892).

[9] L. Cheslock, "Some Notes on Perfect Pitch," The Am. Merc. p. 459 (1934).

[10] C. Stumpf, *Tonpsychologie* (S. Hirzel, Leipzig, 1883–90), second volume.

[11] G. Révész, *The Psychology of a Musical Prodigy*, International Library of Psychology, Philosophy and Scientific Method (Harcourt, Brace and Company, 1925).

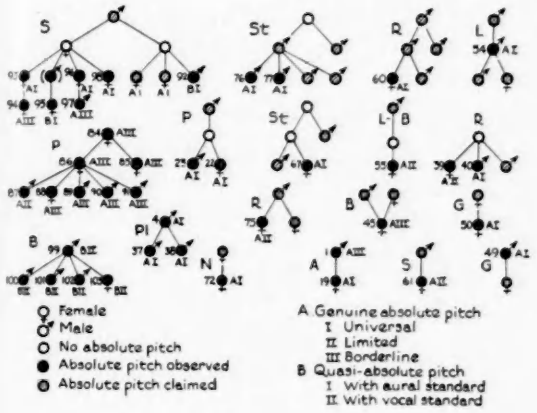

FIG. 1.

pitch of a prodigy, three years old, who inherited his musical talent from his parents.

About one-third of the subjects examined in this study consider their faculty of absolute pitch as "inherited or inborn," and are convinced that they could recognize and identify tones as soon as they learned the musical scale.

One blind person without musical training detected his absolute pitch accidentally during a conversation with a possessor of absolute pitch, and he was able immediately to identify tones correctly after he knew how to denote them.

Five musical prodigies, $4\frac{1}{2}$ to 8 years old, had very good genuine absolute pitch over almost the whole tonal scale, both for the piano and for other instruments. All these children had early musical training, starting between $2\frac{1}{2}$ and $5\frac{1}{2}$ years of age. Most of them observed absolute pitch soon afterward within 2 weeks to $\frac{1}{2}$ year later. In one case, the father claims that his child has absolute pitch at the age of one year. She knew the French nomenclature (on the fixed scale) of piano tones which she heard, although she could not yet pronounce the words correctly. Three of the five prodigies had relatives with absolute pitch. All of these observations on prodigies point to an hereditary factor in the occurrence of absolute pitch.

Forty one of the 103 possessors of absolute pitch had relatives with absolute pitch. Considering the rarity of absolute pitch, this represents a percentage much too high to be explained by a random distribution of absolute pitch through successive generations.

Most of these cases occur in rather large absolute pitch families. Fig. 1 gives an illustration of this condition. This chart is based upon either direct observations or well-founded statements of reliable persons, with all doubtful claims omitted.

Three of these "absolute pitch families" are of particular interest and were studied in detail. Family S: This is an exceptionally musical family. Previous generations were very musical, probably with absolute pitch. Mrs. M. S. —W. deceased (93), had absolute pitch. Her daughter (94) studies music, has absolute pitch (variable, inaccurate). Mr. C. S.—brother of (93)—had absolute pitch, but lost it due to musical inactivity. His daughter (95) is a music student, playing violin, piano, and singing; she has quasi-absolute pitch with violin A as standard; her absolute pitch has improved within two years. Mrs. E. S.—P. (96) is a music teacher, with finest absolute pitch, practically infallible. The son (97), 17 years old, has some absolute pitch, inaccurate, variable. Mrs. E. S.—N (98) is an orchestra conductor and music teacher and has good absolute pitch. A cousin of the three sisters (92) has absolute pitch also and claims that his two sisters in Sweden have absolute pitch. It is evident that in this family the grade of absolute pitch is high in one generation but poorer in the next. It is the only family in which two different types of absolute pitch occur.

Family P: 8 members of this family, representing three generations, Mrs. P (86), her mother (84), her identical twin sister (85), and the children (87–91) (4 sons and 1 daughter) have the same type (A) and similar grades (II, III) of absolute pitch; genuine, but inaccurate, with some recognition of key. The mother's key recognition is better than individual tone recognition. In all others it is the opposite. The boy of 17 years has the best (AII), the one of 8 years the poorest absolute pitch. Their father has no absolute pitch.

Family B: 5 members (father (99) and all his children (100–103), 3 sons and 1 daughter) have the same type of absolute pitch (BII). It is better in the children who had good musical training than in the father who had very little training. The mother comes from a very musical family, but has no absolute pitch.

In the two latter families and in the others studied (as close as it could be determined) the type of absolute pitch was identical for the members of each family, although the grade varied occasionally.

All this evidence suggests the existence of an heriditary factor. An alternate possibility could be that the musical environment surrounding these families played an important role. This factor can be ruled out in two families, where the grandfather had absolute pitch and the mother was not very musical. The two musical prodigies (22) and (23) studied on a piano which was so much out of tune that they had to distinguish between the name of the key and the actual tone produced. Under such conditions it would seem

more likely that the development of absolute pitch would be inhibited than promoted.

The influence of musical environment on the acquisition of absolute pitch is rendered very doubtful by the rarity of the occurrence of absolute pitch despite the musical education and the musical interest of the general public, and by the relatively small percentage of absolute pitch in orchestra players.

III

That inheritance is not the only factor involved in the creation of absolute pitch is shown by the abnormally large percentage and the high degree of absolute pitch in congenitally blind persons. Eleven of the 103 cases studied were persons blind since birth; there were 5 more, which could not be reached for observation. Assuming that there are about 2000 blind persons in Chicago (a figure based upon reliable information), it is evident that the blind furnish an exceptionally great percentage of possessors of absolute pitch. They represent a well-characterized group. Most of them had musical training rather late, around the age of 10 years, but they showed evidence of absolute pitch long before that time. This group contains an unusually great number of cases with good absolute pitch, in fact the two best cases that have been encountered. Not in a single case was a relative known with absolute pitch; brothers and sisters with vision had no absolute pitch. These facts point strongly toward the inference that absolute pitch can develop, if particular attention is given to sounds, as naturally occurs more in blind than in seeing persons.

This influence of forced attention to sound by the blind became evident also, when a man was examined who had lost his vision at the age of 7 and who was supposed to have absolute pitch. Results showed, however, that he had the most remarkable relative pitch. Long discussions or wild modulations on the piano could not disturb his memory for tones heard previously, and his interval sense led him to very correct judgments. But when the reference tone was lost, he was helpless, and all his judgments were slower, less certain, and more often off a half-tone than with possessors of absolute pitch.

The preponderance of absolute pitch ability in the blind is illustrated in Table III.

While these statistical results are based on rather incomplete figures and are not strictly comparable for the various groups, they demonstrate the greater prevalence of absolute pitch among the blind than in the normal, in the non-musical as well as the musical groups.

TABLE III.

<table>
<tr><th colspan="3">BLIND PERSONS</th><th rowspan="2"></th><th colspan="3">SEEING PERSONS</th></tr>
<tr><th>ABSOLUTE PITCH</th><th>TOTAL GROUP</th><th>PERCENTAGE</th><th>ABSOLUTE PITCH</th><th>TOTAL GROUP</th><th>PERCENTAGE</th></tr>
<tr><td>11 studied, 5 more known; probably still more existing</td><td>about 2000 blind persons</td><td>about 1%</td><td>Total population</td><td>103 studied, 20 more known; probably still more existing</td><td>about 4,000,000 people in Chicago, suburbs and Urbana</td><td>0.0003% or higher</td></tr>
<tr><td colspan="3"></td><td rowspan="5">Professional musicians</td><td colspan="3" rowspan="5">5%</td></tr>
<tr><td>6 members of the blind</td><td>36 members Choral Club</td><td rowspan="3">14%</td><td>7 students in the school of Music, University of Illinois, Urbana</td><td>146 students</td></tr>
<tr><td>1 member of the Chicago Tuners Association of the Blind</td><td>12 members</td><td>6 Children in the American Conservatory of Music, Chicago</td><td>60 Children</td></tr>
<tr><td colspan="2"></td><td>about 8 musicians in Chicago Symphony Orchestra and Chicago Women's Orchestra</td><td>200</td></tr>
<tr><td colspan="2"></td><td>1 member of the Piano Tuners Association of Illinois, Chicago</td><td>55 members</td></tr>
</table>

The chart also shows that absolute pitch occurs at a decidedly higher percentage in musical groups, particularly professional musicians, than in the population at large. This is particularly true for prodigies and outstanding musicians with early musical training. It seems that attention to musical tones in early youth (as in the blind, and in many musicians) plays a predominant role in the development of absolute pitch. In later years, absolute pitch cannot be acquired, despite all efforts.

IV

Musical experience, however, plays a role in the acuity of absolute pitch ("keenness" in the terminology of musicians).

Several prodigies were examined twice in an interval of one year and although their absolute pitch was very good in the beginning, a small improvement was noticeable concerning the avoidance of half-tone errors in the highest part of the musical scale.

While most subjects examined were convinced that their acuity of absolute pitch was always the same, several believed that musical experience improved their quality of absolute pitch, and lack of experience allowed it to decline. One member of an absolute pitch family asserted that he possessed absolute pitch in his childhood, but had lost it through musical inactivity.

Two musicians found their absolute pitch declining with increasing deafness, which had not progressed sufficiently, however, to interfere with their musical activities. Many nonprofessional musicians and many European immigrants are disturbed by the different pitch standards of their own piano, the European standard being $a=435$ dv and the American Universal pitch $a=440$ dv, with the result that half-tone errors occur in the recognition of single tones as well as in the key identification of unknown music.

A somewhat greater acuity of absolute pitch was found in violinists than in piano players. Two violinists with the greatest acuity attribute this fact primarily to the attention to proper tuning, and believe that the fine distinction between flat and sharp and leading tones contribute to this keenness. All this evidence shows that the acuity of absolute pitch depends to a large degree upon musical experience.

V

From all these considerations it seems that genuine absolute pitch cannot be acquired in adulthood and unless a certain predisposition exists, it cannot be acquired in early life either. This predisposing factor, or the ability to acquire absolute pitch, occurs in highly talented musicians (prodigies, conductors, composers, and concert pianists) with decidedly greater frequency than in average musicians (orchestra players, piano teachers, and piano tuners). There is evidence that heredity plays an important role in this ability factor, just as it does in other factors of musical and other talents. Under the abnormal conditions of those blind since birth, the importance of this predisposing factor is overshadowed by the extreme attention to sound in early life, which develops the acoustic faculties of blind persons to a far higher degree than those of the normal, so that they may acquire absolute pitch.

When genuine absolute pitch exists, its extent and accuracy can be improved by experience; it may also decline by lack of such.

It is evident then, that inheritance, attention and experience are the important factors for the creation of genuine absolute pitch, inheritance being of prime importance in the talented musician, attention representing the most important item in the blind, and experience determining the degree and extent of the final product, particularly in the professional musician.

A Note on Just Intonation

LLEWELYN S. LLOYD
Department of Scientific and Industrial Research, London, England
(Received February 10, 1940)

IN Part IV of his Faculty Research Lecture at the University of California at Los Angeles on May 6, 1936, on "Some Cultural Applications of Modern Acoustics," Professor Vern O. Knudsen discussed the relation between acoustics and music. He said: "An epochal examination was made by Helmholtz, and is recorded in his monumental *Tonempfindungen;* but this dates back to 1862. Since then modern acoustics has given us vastly superior research tools, by means of which many new and significant facts of hearing have been discovered. Inasmuch as the ear is the instrument by which we must judge what is harmonious and beautiful in music, we should search more fully for its secrets if we would best know how to please it."[1] The last sentence sums up the whole problem of musical acoustics; for, as Helmholtz reminded us, by leaving the ear out of the picture theoretical writers of earlier date had reached conclusions which were contradicted by the practice of composers: "We must distinguish carefully between composers and theoreticians" (*musikalischen Theoretikern*).[2] It recalls the observation that Helmholtz set out at the beginning of his *Tonempfindungen:* "Now whilst the physical side of the theory of hearing has been already frequently attacked, the results obtained for its *physiological* and *psychological* sections are few, imperfect, and accidental. Yet it is precisely the specially physiological part—the theory of the sensations of hearing—to which the theory of music has to look for the foundations of its structure."[3]

Helmholtz himself made outstanding contributions to our knowledge of physiological acoustics. More recent work has filled some of the gaps in our knowledge of *aural perception;* and the results of a great part of it have first appeared in the pages of this journal. But much more remains to be done; and because there is embedded in music itself, and in the history of musical composition, much practical evidence concerning the perceptions of the musical ear, I venture to submit some comment on it for the consideration of other members of the Society who may be interested in musical acoustics. How useful this evidence may prove, in checking the conclusions of further investigation, it is not possible to foresee. The difficulty lies in extracting it from music, and the task calls for scholarly cooperation between the man of science and the musician. The first condition of this cooperation is that each shall learn to understand the language used by the other.

If we look back, across three-quarters of a century, to the time when *Tonempfindungen* first appeared, we find a world of scientific knowledge, of musical theory, and of music itself which is very unlike that we know today. Everyone of us will agree that, in the interval, the world of scientific knowledge has altered out of all recognition, in acoustics no less than in atomic physics. Not all readers of this journal, however, may know how completely the musical world in which we can move today differs from that known to Helmholtz's contemporaries. Helmholtz's musical outlook was that of what musicians call the classical period; and since this period one new chapter and part of a second have been added to musical history. But that is far from being the whole story. Now that the classical period has long been ended, it is easy to discern it as being only a chapter in musical history. There were earlier chapters each of which, like that of the classical period, came to an end with great composers who summed up the achievement of a musical generation. Seventy-five years ago, these earlier chapters had been partly or wholly forgotten. J. S. Bach had ended one such chapter. His rediscovery began with Mendelssohn. I well remember the interest which the reappearance of more and more of his cantatas caused in the University of Cambridge (England) forty or more years ago. A chapter which is even more significant, acoustically, was that called by

[1] V. O. Knudsen, J. Acous. Soc. Am. **9**, 180 (1938).
[2] H. v. Helmholtz, *Tonempfindungen* (Eng. trans. of 1875), p. 345.
[3] Reference 2, p. 5.

musicians the polyphonic period. One of the most exciting musical events of the present century, in England, has been the rediscovery, through the devoted services of a small band of scholars, of the treasures of our Tudor church music. The polyphonic period came to an end with great composers like Palestrina (1525–1594) and Byrd (1543–1623). The music of this period achieved perfection at their hands: it has left an indelible mark on later music, but it is essentially different. It is, perhaps, because it is nearer to the beginnings of occidental music that it is so instructive for those of us who are interested in the relation between music and acoustics.

This remarkable chapter in musical history was almost a sealed book to Helmholtz's contemporaries; and what they knew of it they viewed through the eyes of classical tonality. The consequence was fortunate for their ideas of musical technique. Theoretical writers of Helmholtz's day naturally accepted as authoritative the academic musical opinion which was then current. That opinion had lost sight of the contrapuntal origin of contemporary music, and was engrossed in its harmonic aspect. Equally was it obsessed with the then-current influence of tonality; with the idea of a major or a minor key, and the relation of the musical "notes"[4] of the scale to the keynote.

Let us go back to origins. The early European music was melodic. It flowered and came to fruition in medieval plainsong, which was modal. But that could not content men for all time; and the possibilities of concurrent melody engaged their attention. At first this had moved in parallel lines, in what was called "organum"; but by the thirteenth century, when *Sumer is icumen in* was written, the art of music had decided that the future lay in diversity and some independence of movement.[5] (It was as a corollary that consecutive fifths and octaves later became taboo.)

There is considerable variety in the melodic scales evolved by different peoples; and in a given melodic scale there is little rigidity of intonation. It is a fortunate thing for occidental
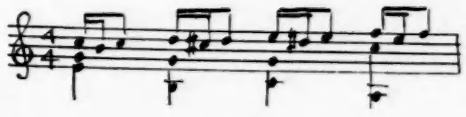
FIG. 1.

music that the modes used in medieval plainsong lent themselves so readily to concurrent melody, which became polyphony. But we cannot assume that the scale of plainsong was the scale we should discover in the concords of the polyphonic period. It is interesting to find that Thomas Morley, in *A Plaine and Easie Introduction to Practicall Musicke* (1597) describes the scale (of plainsong) as having what we should call Pythagorean intonation. He records that, as a result, the semi-tone would be too small by a comma.[6] Observe, then, that the possibility of such a melodic scale was not questioned by this famous madrigalist; and a madrigal is pure polyphony. The difficulties of theorists were of later date. That string players tend to produce a similar intonation, in unaccompanied melody, was established by Cornu and Mercadier, seventy years ago. Nor does the development of music in the interval appear to have caused any change of sensibility in the violinist's ear. P. C. Greene has shown, so far as can be shown by measuring the frequencies of vibrations in air (which are not necessarily to be identified with the intonation perceived), that violinists, *playing unaccompanied*, use quite a flexible intonation, which tends to approach Pythagorean intonation as a "norm."[7]

There is nothing surprising to musicians in this. A violinist playing the top line of music in Fig. 1, even when supported by the harmony shown, would tend to close the decorating semitones, particularly those using chromatic notes, so as to make them less than even the semi-tone of equal temperament. They might well become, as nearly as may be, the half-tone of Pythagorean intonation. Let us combine this musical fact with the masterful importance of the perfect fifth in all intonation, melodic or harmonic. It becomes evident that an intonation approximating to that

[4] The quotation marks indicate that the term is quoted from the writings of English musicians mentioned below, and with the sense in which they use it.

[5] *Encyclopaedia Britannica*, 14th ed., article on "Harmony," Tovey.

[6] On the second page of the "Annotations" at the end of the book, now available as a reprint: "Shakespeare Association Facsimiles No. 14."

[7] P. C. Greene, J. Acous. Soc. Am. 9, 43 (1937).

of the Pythagorean "scale" will often be felt, by the violin player, to be the "natural scale" for unaccompanied melody. On the other hand, his actual intonation will be largely influenced by the musical figure he may be using.

Now suppose three or four performers to be singing concurrent melodies, and that they want them to sound harmonious. They each tend to employ a flexible melodic scale. But they will evolve, as composers of the polyphonic period evolved, a system by which concords will be sung in conformity with some plan; for example, on what we should call the accented beats of the measure. Those concords they will wish to hear *in tune*. Let us think of this in the light of Helmholtz's work and his theory of consonance. The intonation of the intervals that form the concords will be given an exactness which will depend on the nature of the interval, and the time it lasts; factors which, as Helmholtz showed, give the interval its definition.

This very imperfect and brief account will perhaps serve to explain what the counterpoint of the sixteenth century means acoustically, for by counterpoint we mean concurrent melody. Musically its outlook is horizontal, as opposed to the vertical view of harmony. Scientifically, it brings the time factor into the picture. The purest consonances, the octaves and fifths (the perfect concords of counterpoint) are sharply defined. The imperfect concords, the thirds and sixths, will be less sharply defined; but Pythagorean thirds, between tones heard together, will be so faulty as to sound out of tune to a delicate ear. The discordant intervals, such as the dissonant tritone, will have far less definition than the concords. That is why enharmonic change, say the substitution of $F\sharp$ for $G\flat$, became possible at a later stage as a deliberate musical effect.[4] (The interval C to $F\sharp$ is the tritone of the key of G major.) Finally we have "notes" which may not be part of the prevailing harmony, and these will enjoy much of the freedom of intonation of melodic scales.

"Notes" which are not part of the prevailing harmony the contrapuntist describes as "unessential notes"; and examples have been marked with an asterisk in Fig. 2.[5] Their intonation will
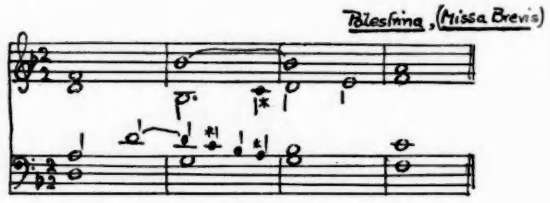
FIG. 2.

be governed by the musical instinct of the artist. It is rather a remarkable thing that in nearly all discussion of intonation, by theoretical writers (Moritz Hauptmann is a notable exception, but then he was a violinist), the existence of these "unessential notes" in music is ignored. This is no doubt the result of the "miasma" described by Sir Charles Stanford in a passage quoted below. We may think of the "unessential notes" as pursuing their own way, unfettered, with an intonation that depends, for its realization, on the artistry and the skill of the player. But when a concord on a beat of the measure is reached, it will always try to pull the intonation of the several parts into a harmonious relationship. It is, indeed, precisely in this fact that the peculiar pleasure of singing in an unaccompanied partsong lies.

This attempt to give a potted history of harmony is very crude and inadequate; but it is perhaps excusable if it serves to show why the approach to music through counterpoint presents a picture which is wholly consistent with that presented by physiological acoustics and the theory of hearing. For as Bernard van Dieren observed: "The alleged problems of euphony that obsessed theorists solve themselves in well-balanced polyphony."[9]

Further significance of this story, for acoustics, lies in its bearing on the meaning of a musical "scale." Composers do not think, nor do artists play, *in* a musical scale, if by scale we mean a series of intervals of fixed intonation. The composer simply writes music. If we want to discover the nature of the material he uses, we must distill his scale from the music he writes. Failure to

[5] Figure 2 also illustrates the procedure so characteristic of Palestrina: after moving by *leap* a voice generally returns by *step* in the direction from which it has come. Acoustically the point is worth noting because it helps the singers to maintain pitch, see *Musical Composition*, C. V. Stanford, p. 127. A full examination of this procedure for leaps of different intervals will be found in *Contrapuntal Technique in the Sixteenth Century*, R. O. Morris, pp. 29 and 30.

[9] B. van Dieren, *Down among the Dead Men*, p. 226.

realize this has been responsible for more mistaken theory about music than anything else. That is the reason for the warning given by Sir Hubert Parry, sometime Professor of Music in the University of Oxford, but better known as an English composer: "It is advisable to guard at the outset against the familiar misconception that scales are made first and music afterwards. Scales are made in the process of endeavouring to make music."[10]

I have tried to set out this story, as far as possible, in terms of our knowledge of acoustics: but now let us turn to a distinguished teacher of composition. In Chapter II of his *Musical Composition*, Sir Charles Stanford discusses technique with the beginner. Here are passages:

"The simultaneous presentation of two melodies which fit each other is at once a musical invention; and when a third and a fourth melody is added to the combinations, the result is what is called harmony. To speak of studying harmony and counterpoint is, therefore, to put the cart before the horse. It is counterpoint which develops harmony, and there is no such boundary-wall between the two studies as most students imagine. . . .

"To begin technical training with harmony gives rise to a habit in a beginner, which it is most difficult to eradicate when he embarks on composition, the habit of harmonizing every note of the melody. . . . [Cf. what was written above about unessential notes.]

"The first principle to be laid down is, therefore, to study counterpoint first, and through counterpoint to master harmony. . . .

"The policy of putting harmony before counterpoint is of comparatively recent growth; the growth has unfortunately overrun a great deal of low lying land, and it is easy enough to note where it flourishes from the results of its miasma."

The reader who now turns to Sir Donald Tovey's article on "Harmony" in the fourteenth edition of *The Encyclopaedia Britannica* will find the same story told in a somewhat different setting, with a racy dismissal of the English theory of harmony in vogue in Helmholtz's day as "Dr. Day's famous application of homoeopathy to the art of music" (Dr. Day practiced homoeopathy in London in the first half of the nineteenth century.) And, if he recalls what Helmholtz teaches us of the way in which the ear apprehends a consonance, he will appreciate that the advice given by Stanford and Tovey is good science as well as sound musicianship. That is why the opinions of these distinguished musicians are so instructive for students of musical acoustics. We shall all agree that the first duty of a man of science engaged in investigation is to examine his data; and he will accept as an axiom that, to understand the relations between acoustics and music, we must begin by stating our musical premisses correctly.

All of which brings us to our subject. "Just intonation," "the pure scale," "the natural scale" have often been used as synonyms for the tuning of a keyboard instrument in a manner which gives perfectly tuned triads on the tonic, dominant, and subdominant.[11] This has two unfortunate results. First, it will obscure to the reader the meaning which writers like Stanford and Tovey attach to these terms. Second, it will mislead them about the meaning Helmholtz attached to "just intonation."

The tuning of a keyboard instrument has an arithmetical basis, because it is concerned with rates of vibration which are measured by beats. An arithmetical approach is therefore appropriate in the discussion of tunings intended to represent, as well as may be on a keyboard instrument, the music of particular periods. But when we are investigating the scale which is implied in music as conceived by the composer, and rendered by a good artist on an instrument with free intonation, we must begin with hearing, not with arithmetical relationships. The true musical approach is that adopted by Professor A. T. Jones in his *Sound:*

In the key of *c* major let us see what the scale will turn out to be if we use no other notes but those in the successive major triads *f–a–c*, *c–e–g*, and *g–b–d*.

We turn to Helmholtz to discover *why* the intervals of these triads will approximate to those between musical sounds produced by vibrations in the ratios 4 : 5 : 6, the problem which, as he explains, he set himself to solve in the second

[10] Sir Hebert Parry, *The Art of Music*, p. 16.

[11] I.e., *c*, *g*, and *f*, in the scale of *c* major.

part of *Tonempfindungen*.[12] He solved it by explaining how the musical ear tells whether two tones heard together are "in tune." Then, following the argument used by Professor A. T. Jones, we obtain a succession of tones which will give ideally true intonation for what the musician calls a perfect cadence and a plagal cadence, the progressions which are familiar to us as the two forms of Amen as sung in the Liturgy (Fig. 3):
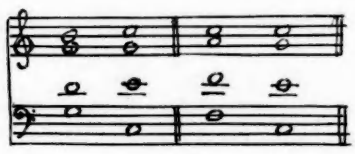
FIG. 3.

Music does not consist, however, of the wearisome reiteration of tonic, dominant, and subdominant harmonies, which is, for example, the first thing the music student has to avoid in learning to write "canon at the octave"; for these cadences, in music, are the equivalent of full stops. Harmony on the second note *d* affords relief. But the interval between *d* and *a* obtained in the tuning we are considering is not a perfect fifth. It is too small by a comma. On page 62 of *Sound*, Professor A. T. Jones reminds us that Mr. Redfield proposed a new scale (i.e., "tuning") in which the interval from *c* to *d* would be "represented by 10/9." Such a tuning would produce a perfect minor triad *d–f–a*, but the interval *g–d* in the first concord of Fig. 3 would now be out of tune. It would be too small by a comma.

The truth is, that in the pure scale of strict counterpoint, as Stanford points out, *d* becomes quite unstable under the varying stress of different concords, and has to move about. It becomes a "mutable note."[13] Helmholtz drew attention to the same fact, and added that in the diatonic scale of *c*, not only must *d* be prepared to be either a major tone or a minor tone above the tonic *c*, but *b*♭ must be prepared to be either a major or a minor tone below the octave *c*.[14] The same is true of a minor scale. Thus in the key of *a* minor, *b* and *g* (of the descending melodic minor) may be separated from *a*, or its octave, by either a major tone or a minor tone; while *d* for the subdominant will occupy the place assigned to it in Mr. Redfield's tuning.

While the unstable quality of at least two "notes" of both the major and the minor scale of music is incompatible with the rigid intonation of a keyboard instrument, the instability of the "notes" of modal music was even more extensive.

In an article which will shortly appear in *The Musical Quarterly*, I have examined the intonation of the opening strains of Taverner's *Western Wynd* Mass, assuming as my musical premiss that the concords are heard as being sung perfectly in tune. The result will perhaps be instructive for those readers who wish to follow further the method adopted by Stanford to derive what he called "the pure scale" from modal counterpoint and the polyphony of our Tudor composers.

Observe, then, that by "pure scale" Stanford does not mean merely the intonation required for ideally perfect triads on the tonic, dominant, and subdominant. He is thinking of the intonation of the concords in contrapuntal writing; and we must never forget that this does not settle the intonation of what have been described as "unessential notes"; though as a concord is the goal towards which the unessential notes are leading, it will to that extent have some influence on their intonation. The student of counterpoint who reads *Tonempfindungen* (if in the English translation, carefully ignoring the translator's additions) will lay it down with the conviction that, notwithstanding Helmholtz's limited acquaintance with sixteenth century music and his acquiescence in Rameau's harmonic doctrines, his outlook was instinctively that of a polyphonist. This becomes especially evident in occasional passages such as:

> Different voices, executing the same or different melodic phrases, can move at the same time within the compass of the scale, like two bodies in space, and, provided they are consonant in the accented parts of the bar, without creating any musical disturbance.[15]

When musicians began to use such keyboard instruments as virginals, in the sixteenth century, they inevitably encountered the fundamental difficulty that no succession of fifths, or major thirds, can ever be fitted, exactly, into a succession of octaves. Dr. Fellowes tells us that

[12] Reference 2, p. 24.
[13] C. V. Stanford, *Musical Composition* (Macmillan, 1911), pp. 15 and 16.
[14] Reference 2, p. 423.
[15] Reference 2, p. 576.

there is evidence, in some of the music for virginals, of experimental tunings.[16] But the search for the ideal temperament, for the tuning of keyboard instruments resolves itself into the quest for the temperatment which does least harm to the music for which it is used.

All this will be familiar enough to many readers of this journal. But it is perhaps worth while to go over the ground again, starting from what may be a not-so-familiar point of view, because of the confusion of nomenclature which surrounds the subject. In an article entitled "Just Intonation Confuted" which Dr. Murray Barbour contributed to *Music and Letters* for January, 1938, he described "just intonation" as a perfect example of a "question-begging word." That is quite true, and it is the excuse for this note. English musical writers use the words, as a rule, to denote an intonation which produces perfectly intoned *intervals*.[17] That is the sense in which they are used by Helmholtz; as is evident from the quotation from *Tonempfindungen* which Professor Jones makes on pages 72 and 73 of *Sound*, and from the fact that Helmholtz's justly-intoned harmonium needed two keyboards, as much to cope with "mutable notes" as to permit of modulation. Similarly English writers usually follow Stanford in the use of the term "pure scale" for the flexible intonation required for modal counterpoint, and for all subsequent polyphonic development of the musical scale. It is true that there are English "theoreticians," more especially since the time of Ellis, who have confused the issue by using "just intonation" for the tuning which is described in the Glossary of the British Standards Institution as "just temperament."[18] In the accepted doctrines of academic musical opinion of his day, Ellis had every excuse for his logical, but mistaken, system of "duodenation." The present writer offered a full *apologia* for him in an article on "Intonation and the Musical Ear" which appeared in *Music and Letters* for October, 1938. But Ellis's conceptions were wholly mistaken. On page 73 of *Sound*, Professor Jones quotes from his writings a passage which contains the following:

> "The human voice, the violin, and the trombone . . . will enable the composer to study the wondrous wealth of just intonation as a means of musical expression, when the laws of just modulation have been laid down. . . ."

The idea of composers waiting for "theoreticians" to formulate these laws before they could write music intended to be performed and heard *in tune* is funny enough itself. It becomes absurd when we remember that the textbook "laws" are always at lest a generation behind the practice of composers.[19] Ellis was, indeed, the leading exponent of the misconception that led Parry to make the protest already quoted.

It is perhaps fair to say that, in England, discussion of "just intonation" between musicians and theoreticians has become rather boring; and now attracts little general interest; though it has not succeeded in settling the meaning of the term. If some of us still use it in the sense in which Helmholtz used it, instead of avoiding its use altogether, it is, I think, because we are unwilling to allow the theoretician to get away with his misinterpretation of Helmholtz, or with the misconceptions about the nature of music which Ellis made fashionable and which may still mislead. This confusion of ideas, reflected in confused terminology, is, indeed, of little moment for those who have the time and inclindation to study counterpoint for themselves; not the arid counterpoint of Cherubini, but the living counterpoint of the English and Italian composers of the sixteenth century. If they possess an instinct for music they will learn the musical truth from music itself. But for those who have not this opportunity, and who must rely, in the main, on what is written *about* music by musicians, it is perhaps worth while to illustrate the meaning which is given, by English writers such as Stanford, to the "pure scale," and to explain why we find the perfect example of the question-begging word, used in quite different senses by different writers, in "just intonation."

[16] *The English Madrigal Composers*, p. 98.

[17] Thomas Morley used the word "just" in this sense (see the second page of the "Annotations" to his *Plaine and Easie Introduction*).

[18] *Music and Letters*, **20**, 371 (1939).

[19] Cf. *Tudor Church Music*, Vol. 1, editorial notes p. xl.

The Paradox of Voice Teaching

WILMER T. BARTHOLOMEW
Department of Research, Peabody Conservatory of Music, Baltimore, Maryland
(Received January 19, 1940)

THOSE of us who work in border-line fields are frequently astonished at the amount of misunderstanding which can exist among the specialists who restrict themselves to a single branch of study. Such specialists too often remain ignorant of relevant facts in closely related fields. Down through the decades, for example, the musicians—the makers of music—have been often at odds with the acousticians, not because of any fundamental contradiction between the canons of art and the laws of science, but simply because their respective devotees have not gone deep enough to uncover the common underlying principles. As one example from many, one may cite the use of certain types of piano touch, which may be inefficient from the purely physical standpoint, but of great importance in inducing a physiological relaxation in the player. For that matter, such a movement may not even have a physiological justification, and yet still be legitimate and necessary because of its *psychological* effect on the player or listener.[1] This is a field one more step removed from the physical, and yet no less real because of its subtlety. Where art is concerned, more criteria than those of mechanical efficiency must be considered.

In no branch of music has this lack of understanding been so acute as in the teaching of voice, where the mechanism involved is complex, variable, and never entirely under conscious control. The voice mechanism cannot be examined during tone production except in a very cursory and unsatisfactory way which often creates abnormal singing conditions, and this has by no means lessened the opportunities for misunderstanding between voice teachers and scientists working in the voice field. Anyone who tries to check the statements from one of these camps by reading the literature of the other, will at once be in difficulty. There is of course the customary confusion in terminology, where one group quite literally "does not speak the other's language." But in addition there is often a wide and unnecessary difference in viewpoints.

The results of a program of voice research commenced nearly ten years ago at Peabody Conservatory, checked by comparative studies of the conflicting literature produced by voice teachers, phoneticians, anatomists, surgeons, acousticians, psychologists, and others, have crystallized into a picture of how the voice mechanism probably operates.[2] This picture, although subject to errors through undue simplification, or through the personal bias of the writer, is yet sufficiently accurate in essential details to be used as the norm, the usual operation of the voice in so-called "good production." It is, in brief, as follows:

Good singing is produced by a column of air under steady pressure from the abdominal muscles and intermittent pressure from the chest muscles, which causes the vocal cords to vibrate in either of two types of vibration or, ideally, a combination of the two. In the first type, used for lower pitches, and usually called the "chest register," the cords are relatively thick, and vibrate as a pair of cushions, or as a soft double reed. In the second type, used for higher pitches, and usually called the "head register," the cords are thinned and stretched. Probably only the edge of each cord takes part in this type of vibration, and instead of a complete closure during each vibration the glottis is probably somewhat open all of the time, the cords vibrating more as "air-bowed" strings than as a double reed. The puffs from the cords energize the laryngeal and pharyngeal cavities, and to a less degree the oral and nasal cavities above them, modifying the glottal tone according to the natural frequencies of these cavities, which vary with their volume, the size of their openings, and the character of their surfaces. However, the whole mechanism must be regarded as a unit, since not only does

[1] Otto Ortmann, *The Physical Basis of Piano Touch and Tone* (1925); and *The Physiological Mechanics of Piano Technique* (1929).

[2] W. T. Bartholomew, "A Survey of Recent Voice Research," Proc. Music Teachers Nat. Assoc. (1937).

the glottis affect the resonators, but these react back on the glottis vibration. "Good production" in speech might mean *efficiency*, or in other words how rapid the articulation can be made without losing any of the sounds essential to the meaning. Here the consonants are the important thing. In singing, however, and frequently in speech also, "good production" has an aesthetic import. We apply it to *beautiful* vocalization; and vocal beauty, at least in Western civilization, has come to mean largely the presence of three things: (1) sustained vowels, containing (2) the proper smooth vibrato, and (3) a strong low formant in the general neighborhood of 400–600. When the setting of the resonators that gives the best quality of tone is secured (and this is principally by enlarging the lower throat so that its natural frequency drops from perhaps 1000 in the case of a very throaty voice, to between 400 and 600), the voice is said to be properly "placed," and on this basic quality are superposed the finer qualitative differences responsible for vowel differentiation. Also, when the throat is enlarged as in good singing in the "chest register," a high formant appears, produced by a type of resonance in the semi-cavity formed by the larynx from the glottis up to the aryteno-epiglottic rim. Scattered high frequencies have been reported in the voice by various investigators who used unselected voices. However, when voices are used which by accepted artistic standards are satisfactory, in other words, any artist before the public, or any promising student, these high frequencies will be found rather narrowly limited to a range for men between roughly 2400 and 3200, and frequently averaging at 2800 or 2900, around *e* or *f* of the last octave of the piano, and for women a trifle higher.[3] (Sopranos frequently do not show it since they use principally the "head register," a much less complex tone. It is to be found in all good contraltos, however.) This formant frequently weakens in intensity or even vanishes at some point in each vibrato cycle, but it tends to remain within this same pitch range, regardless of the vowel, the fundamental pitch, or the voice range (bass, baritone, tenor, alto). Curiously enough, this range is the same as the natural resonant pitch of the ear canal.

Here we have, then, a picture of the operation of the voice mechanism, based on a comparative study of a large amount of research, and which can be still further simplified to the two words—*large throat*—for a large throat tends to promote or permit all the attributes of good quality. But when we undertake a comparative study of vocal *method* books, written by the teachers who struggle daily with vocal problems and cast about for remedies and explanations thereof, we find quite a different story. One who has had any experience with voice teachers and their texts is usually so struck by the pseudo-science and the points of disagreement, as to overlook a certain extremely significant point of likeness, probably the only thing common to all methods, namely, the insistence on "getting the tone *up*," somewhere or other. This idea of "upness" is even present in some form or other in methods which stress "singing down on the diaphragm" and similar imagery. The precise position of desired "focus" varies, but they all want something or other *up* somewhere, out of the throat, as they say, so much so that an upward-outward movement of the hand is a characteristic gesture of voice teachers. And right here is where most scientists lose all remaining patience with voice teachers, and voice teachers with scientists. Here is one group insisting that the principal necessity for a good voice is for it to be resonated in a large throat, while another group, however else they disagree among themselves, insists that the principal necessity in voice teaching is to "get it up out of the throat." This contradiction is only apparent, however, since a real psycho-physiological problem underlies it.

There seems to be no doubt that the most important task of the voice teacher, as in fact some have realized, is to teach the student to sing with a large open throat, maintaining this position through as many sounds as possible, and for as high a pitch range as possible. This setting of the throat, secured by relaxing the tongue, jaw, and constrictors, and by enlarging the throat by permitting the larynx to be lowered and the sides of the throat to widen, as in the *beginning* of a yawn or in a deep inspiration, is of the greatest importance. In addition to securing a

[3] W. T. Bartholomew, "A Physical Definition of 'Good Voice-Quality' in the Male Voice," J. Acous. Soc. Am. **6**, 25 (1934).

large pharyngeal resonator it permits more powerful tones to be produced, and encourages the vibrato and the production of the high formant, the "ring" in the voice. However, it is for many a difficult coordination to secure and to maintain, for at least two reasons. First, the large throat runs into constant conflict with the necessity of producing consonant closures. Second, the large throat involves the relaxation of the powerful swallowing group of muscles, which in many individuals are normally in a state of partial contraction, ready to act in the semi-reflex of swallowing. Of the dozen or more muscles which coordinate in the swallowing act, the two pairs known as the pillars of the fauces are perhaps the most dangerous to good voice quality. These are the *palato-pharyngeus* and the *palato-glossus* muscles. Both tend to raise the larynx and thereby make the throat smaller. The former, the posterior pillars, which aid the small laryngeal muscles to stretch the cords to secure the higher pitches, by their powerful operation also diminish the throat size laterally, by closing in at the back of the throat. The anterior pillars, the palato-glossus muscles, tend to lift and tense the tongue, and thereby furnish a fixed fulcrum for the operation of the hyo-glossus muscle, which muscle then tends to lift the hyoid bone, and through the hyoid the attached thyroid cartilage. However, the palato-pharyngeus and palato-glossus muscles must themselves operate from a fixed fulcrum, the soft palate or velum, and if this member is not fixed from above (in other words, if the elevators of the palate are relaxed), tension of the muscles suspended therefrom becomes mechanically impotent to raise the larynx, if indeed not neurally impossible due to the coordinated innervation of the whole group. If the small constricted throat which results from a high larynx and partial contraction of the constrictors and other swallowing muscles is to be avoided, it would appear that the ability to relax the muscles which elevate the palate would go far toward avoiding it.[4] In nearly all of the speech sounds of many persons these palate-raising muscles are tensed, holding the velum up more or less tightly against the rear wall of the throat column, and closing off the entrance to the nasal pharynx. An exception is found in the Italian language, in which the nasalization required for a "nasal" tends to spread to neighboring vowels and sometimes to the entire word. The way that the velum action is bound up with the rest of the swallowing coordination may easily be demonstrated. Breathe naturally in and out through the nose with the lower jaw relaxed. Now tense the tongue muscles, as in the beginning of the swallowing act. If the breath does not immediately stop completely through velar closure, it will at least become audible as it pushes with difficulty past a constricted opening at this point. The elevators of the palate (*levator palati* muscles) have tensed to provide the necessary fixed fulcrum for the muscles below them. Reference to any standard work on anatomy will further clarify this matter.

Actually, these elevators of the palate furnish the most valuable key toward gaining conscious inhibition and control of the automatic swallowing coordination, which is practically reflex in its operation. The elevators of the palate are perhaps the only link in the strong muscular chain that is weak enough to be surprised into relaxation whenever we wish, by a setting which is almost equally "reflex" in its operation. This setting is the phonation of a nasal such as *m*, *n*, or *ng*. We have but to think *m*, *n*, or *ng*, and the velum drops, through relaxation of the levator palati muscles, to permit breath to pass through the nose. The ability to keep the passage to the nose open during all vowel sounds, implying as it does the relaxation of the palate-raising muscles and through them the relaxation of the rest of the swallowing coordination, spells the beginning of better vocal quality.

It becomes very evident, then, why the vocal teacher often uses these humming sounds. It is equally clear why he has been at such pains to secure "head resonance," or to get the tone "in the masque," or "forward," or "against the teeth," etc., since the attempt to do these things usually relaxes the velum in order to open the passage into the head, and thus weakens or destroys the fixity of the upper fulcrum from which the pillars and other swallowing muscles operate, permitting the weaker, downward-pulling muscles to enlarge the throat.

Thus we have the curious anomaly of the nose

[4] W. T. Bartholomew, "The Role of Imagery in Voice Teaching," Proc. Music Teachers Nat. Assoc. (1935).

and head passages adding practically nothing in the way of useful resonance, and yet being of paramount importance in securing indirectly a proper setting of the throat resonators. And from this paradox stem most of the misunderstandings not alone between voice teachers and scientists, but among voice teachers themselves. There is no doubt that the tone quality is almost wholly determined without the *acoustic* aid of the cavities above the level of the roof of the mouth. We have performed the experiment several times of getting an experienced singer who can control the back of the throat muscularly, to sing a good quality tone with the velum opened, and then with it closed, cutting out head resonance. When the throat setting is kept large and constant, the resulting tones cannot be surely distinguished by competent listeners. And yet with some students it is dangerous even to say the word "throat" for fear of an involuntary tensing of the constrictors. The indirect concept of "head resonance" is far safer. However, repeated experiments show that although the *attempt* to feel head resonance frequently improves the tone markedly through the physiologic and psychologic effects described, *and is therefore to be considered good pedagogy*, the actual resonating of sound in the head cavities is of very little importance, if any, in the physical production of good quality. The tingling sensation sometimes felt in the head cavities during singing should be considered more as a result of good production than as a contributing cause.

If the various tricks of the trade that voice teachers use to improve quality are analyzed, most of them, if not all, will be found to be devices for directly or indirectly enlarging the throat. The teacher may not be aware of this, but simply knows that such devices get results. Thus the proper taking in of breath will often loosen the interfering muscles, the jaw relaxing more and more, the deeper the breath. The throat will be stretched and the larynx lowered, because the lungs descend slightly during inspiration, drawing after them the trachea. The feeling of the beginning of a natural yawn, or the relaxation felt after its completion, are aids to securing a large throat. If the yawn be forced, or if it has passed into the rigid phase, this aid is of doubtful value. The ability to groove the tongue strengthens an important member of the throat-enlarging set of muscles, although it must not be attempted to keep the grooved tongue-position while singing. The attempt to feel the resonating of the sound in the nose cavities, or to feel the slow passage of the warm breath through the nostrils while singing, will aid throat enlargement through the necessary preliminary relaxation of the palate-fulcrum.

Such aids are primarily for the vowel-quality, the sustained vowel as used in beautiful singing, or to a less degree in beautiful speech. The voice is unique among musical instruments in its presentation of a thought content through words. And words, unfortunately for vocal tone quality, contain consonants. Consonants are produced by narrowings or more or less complete closures at one or more points between the back of the tongue and the lips, and right here is the explanation of the special difficulty of the singer—a difficulty presented by no other instrument and one which goes far toward explaining the dearth of good voices as compared, let us say, with good violinists or good pianists. Since the voice is almost always under the necessity of singing *words*—which means that it must get across to the hearer many consonant sounds—it must constantly be making throat, tongue, or mouth closures at the same time that it is attempting to keep these parts as wide open as possible for the sake of power and quality. These two things, in their very nature, are forever at odds. Frequently the result is a compromise which favors either tone quality, in the case of the singer, or enunciation, in the case of the typical speaking voice. Only occasionally is a singer able to achieve both *desiderata*, and then it is usually accomplished by learning how to make all consonants emphatic but as short as possible, holding off their entrance while keeping the throat enlarged on a vowel as long as possible, pronouncing the consonant rapidly, and immediately opening the throat for the next vowel. But throughout all of the early training of many, if not most students, beautiful quality can be best obtained by not laying too much emphasis on consonants, especially those requiring severe throat constrictions, but instead calling the attention *away* from the throat, securing its enlargement indirectly by the use of some sort of

"up" imagery. There are innumerable variations of this imagery. One teacher will tell the student to "keep the *ng* in all vowels." Another: "Sing as if the head were full of sound." A prominent writer in discussing the training of choir boys, desires "resonance right at the top of the skull," although he does not explain where the grey matter is to be removed to in order to secure the necessary space for resonance to occur. Still another, even more fanciful perhaps, and yet equally effective in securing results: "The vibrations of the voice should always be tasted in the front of the mouth—not gargled in the throat." Any typical textbook on voice will furnish numerous other examples. The variations are legion and often ludicrous.

Thus vocal methods are by no means as incompatible or irreconcilable with one another, nor with science, as if often thought. The differences, often magnified through professional jealousy, are in the type of imagery used, and when methods conflict, the disagreement is almost always to be located, not in the realm of physical fact, but in this irrational realm of imagery, so necessary to the teacher except with a few very intelligent students, while at the same time so delusive and untrustworthy. If teachers will recognize this imagery for what research has shown it to be, a useful tool for securing the far more important but evasive large throat, together with the artistic type of vibrato which usually ensues, let them use whatever type of imagery they wish. But they should by no means believe any particular type of imagery or method to be an indiscriminate panacea for all students. The intelligent teacher will be familiar with all types of pedagogical imagery, including the various synaesthesiae of form, shape, substance, color, or taste, on which we fall back when we try to describe tone qualities, and familiar as well with all the tricks useful for changing the setting of the mechanism, and with the precise effect of various phonetics, and then be analytical and open-minded enough to apply the particular remedy to each particular ill. The scientist, in the meantime, should be open-minded enough to realize that the voice teacher's lingo is by no means all fanciful, but based, perhaps unconsciously, on a curiously complex but very real physiological structure. When in the midst of profound disagreement one point of almost universal agreement is found, a modicum of truth is surely responsible. And the voice teacher's *idée fixe* of upward placement, though an indirect procedure, represents for many students the only practical way of securing a muscular setting which is awkward for the beginner.

Resonant Frequencies and Damping Constants of Resonators Involved in the Production of Sustained Vowels "O" and "Ah"

DON LEWIS AND CURTIS TUTHILL
State University of Iowa, Iowa City, Iowa
(Received January 24, 1940)

IN a paper by Lewis,[1] published in 1936, a method was developed for determining the resonant frequencies and damping constants of the principal resonators typically involved in the production of sustained vocal tones. The method, despite its shortcomings, seems on the whole to be reasonably satisfactory, and it has therefore been used in further investigations.

In its application to a specified sustained tone, the method provides for a Fourier analysis of oscillographic tracings of several individual waves (periods) selected from among the waves within a single frequency-vibrato cycle. The selection of the particular waves to be analyzed depends upon their fundamental frequency, a representative sampling being made. During the course of a vibrato cycle, each harmonic varies in frequency, the extent of the variations in the overtones being proportional to the extent of the variation in the fundamental. In the case of the higher overtones, changes in frequency are quite extensive even when the change in fundamental frequency is relatively slight. Many of the individual overtones are found by analysis to vary in intensity more or less systematically among the several analyzed waves. It is believed that intensity variations of this type are due primarily to the selective effects of essentially unvarying resonators upon harmonics which vary in frequency.

Information on the resonators themselves is secured in the following way. All of the analytical data for a given tone, regardless of the number of waves analyzed, are plotted on a single graph, with intensity and frequency as coordinates. The number of plotted values for each harmonic is always the same as the number of waves analyzed. Mainly for purposes of identifying analytical data for individual harmonics but also to emphasize intensity variations, the several plotted points for each harmonic are connected by straight lines. The resulting composite graph usually shows three or more general contours, each of which tends to resemble the response curve of a simple resonator. By a series of approximations, with due consideration being given to the supposed nature of the cord-tone spectrum, simple resonance curves are fitted to the experimental data.[2] The resonant frequencies and damping constants denoted by these curves are held to be essentially the same as the resonant frequencies and damping constants of the resonators involved in the production of the tone being studied.

The present report deals with typical data on two sustained vowels, "O" and "Ah," as produced by male voices. A study was made of forty-nine tones in all, of which twenty-nine were "O's" and twenty "Ah's." Each tone was produced in a "dead" studio by one of several different trained singers, and the oscillograms were recorded with a system whose response characteristics were known. The analyses were made with a 40-component Henrici analyzer.

Figure 1 gives results on six "O's" as produced at six different pitches by six different singers. In each of the six graphs, relative intensity in db is plotted against frequency. The +20-db point on the ordinate in each graph here (as well as in each graph in Figs. 2 and 3) corresponds approximately to the total intensity of the particular tone represented. The actual analytical data as well as simple resonance curves of best fit are shown. Consider the top graph. Singer B.S., a trained tenor, intoned the vowel "O" at G_2(=196 cycles). Six waves, selected from those within a single vibrato cycle, were analyzed. These waves ranged in fundamental frequency from about 184 to 200 cycles. On the graph, the six plotted values for the fundamental component are clustered around +11 db, just below 200 cycles. Only slight

[1] Don Lewis, "Vocal Resonance," J. Acous. Soc. Am. 8, 91–99 (1936).

[2] See Lewis' paper for a more complete explanation of the method used and for a discussion of the assumptions underlying it.

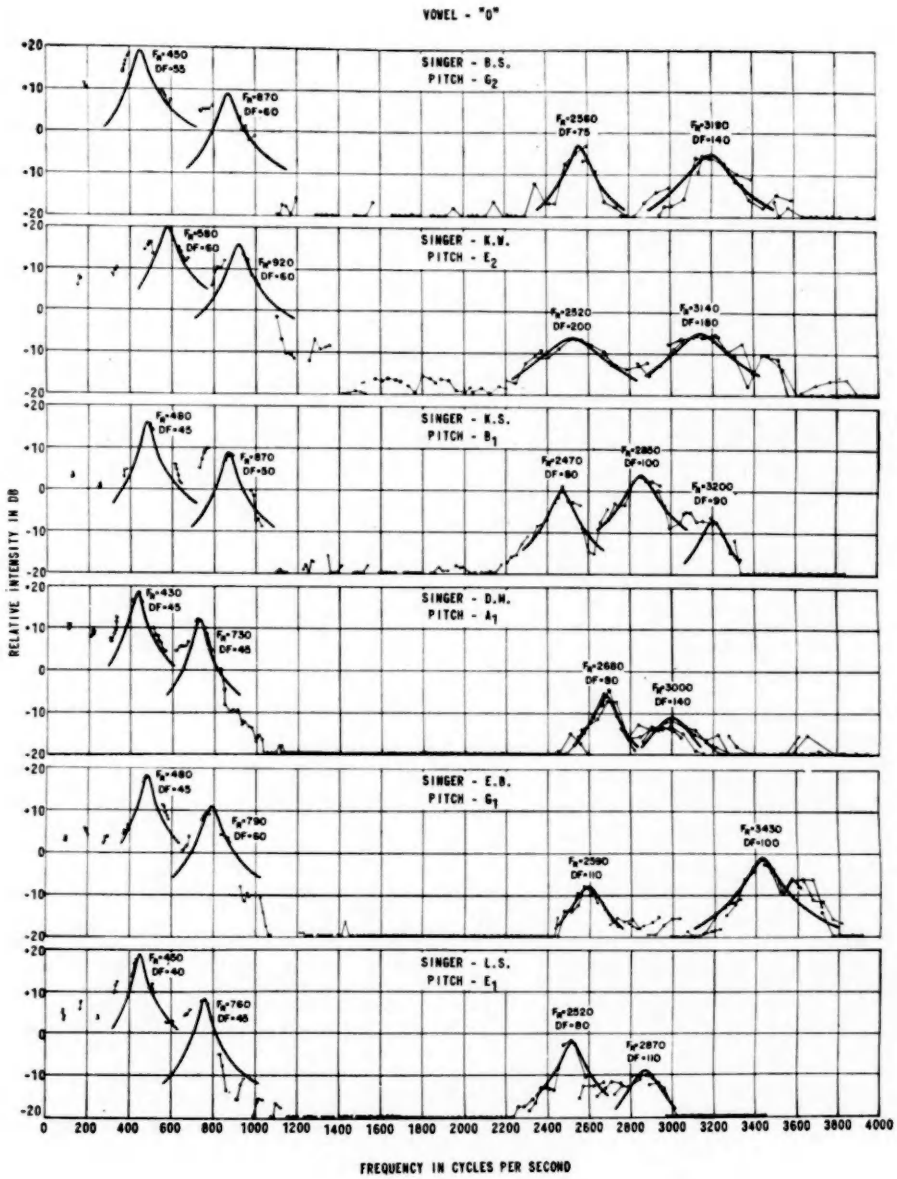

FIG. 1.

intensity variation is indicated. The six values for the 2nd harmonic are shown between about 368 and 400 cycles, the change in intensity being apparent. Values for the other harmonics, up to and including the 20th, are represented. The four general contours as formed by the experimental data may be seen, together with simple resonance curves of best fit. The resonant frequency F_R and the frequency band width DF for each curve are given. Damping constants in db per second may be readily computed by multiplying values of DF by 27.3. There was no energy of importance between 1200 and 2400 cycles or above 3600 cycles.

The other five graphs in Fig. 1 need no specific interpretation. Each one shows at least four general contours and a corresponding number of simple resonance curves, values for the latter being given. Both similarities and differences may be noted among the resonators for the six "O's." With one exception (second graph from the top), the first resonators have resonant frequencies

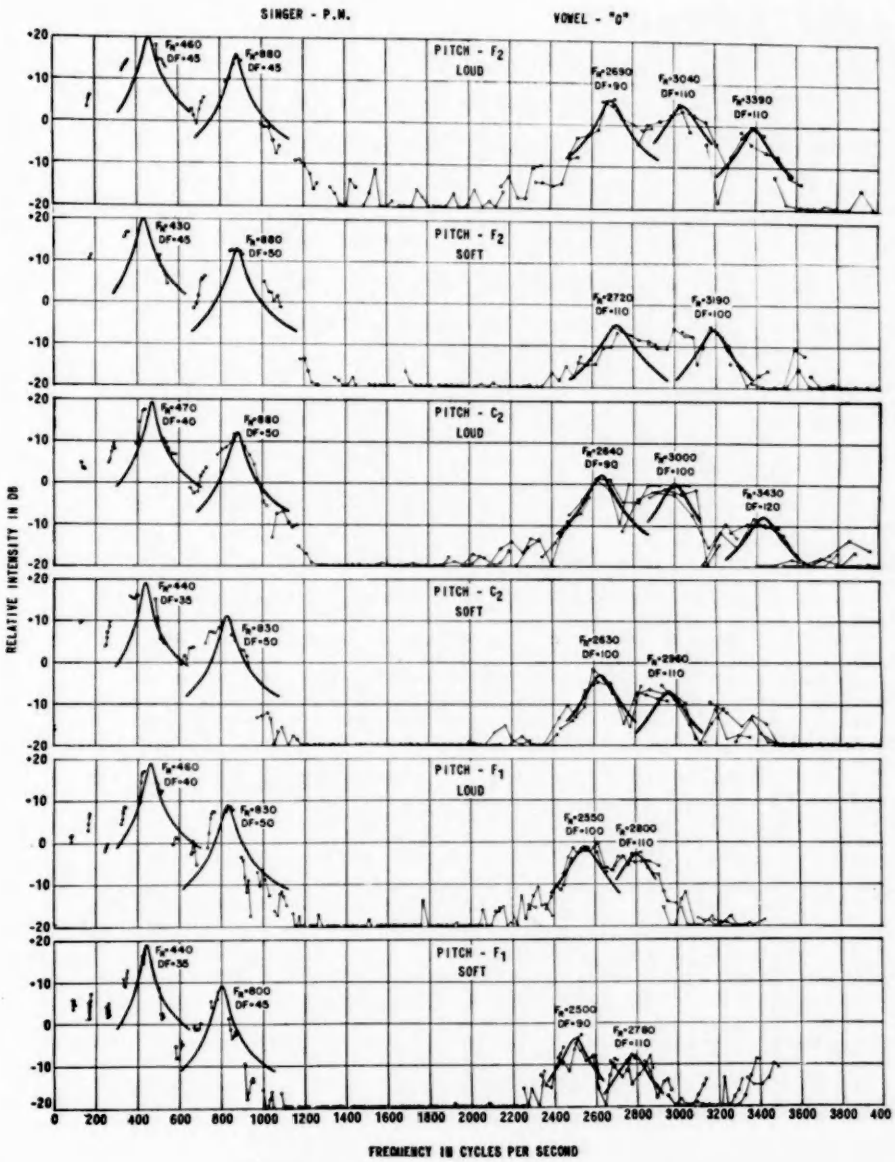

FIG. 2.

fairly close to 450 cycles.[3] Their damping constants range from about 1100 to 1600 db/sec. The second resonators, with damping constants between 1200 and 1600 db/sec., tend to be somewhat less consistent in resonant frequency than the first resonators. The high frequency resonators are more or less unique for each singer although there is always one (the third) with a resonant frequency somewhere between 2400 and 2700 cycles. The damping constants for these high frequency resonators range from about 2000 to about 5500 db/sec. Most of them, however, fall around 3000 db/sec.

Figure 2 presents data on six "O's" as produced by singer P.M., a trained bass. Three different pitches were used, each at two different intensities. The loud and soft tones, at each pitch,

[3] It is suggested, in view of the results obtained on the vowel "Ah," that the tone represented in the second graph from the top in Fig. 1 was probably closer to an "Ah" than to an "O." The experimenter reported the tone as being an "O," but he may have been somewhat amiss in his observations at the time the oscillographic record was made. Unless a singer is quite careful, he will tend to make an "O" progressively more "Ah-like" as he sustains it for several seconds.

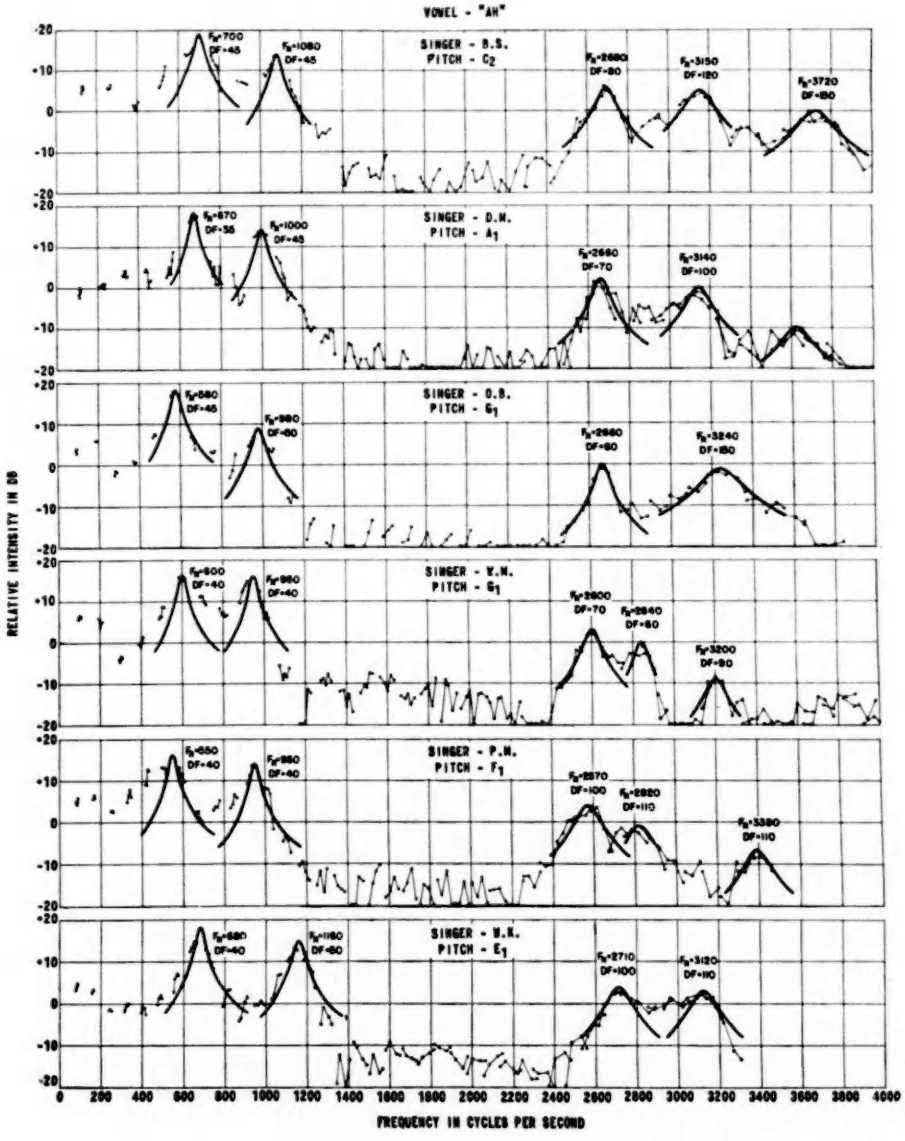

FIG. 3.

differed by about 10 db. There are either four or five general contours in each graph and a like number of simple resonance curves. Values of F_R and DF are, on the whole, quite similar to those shown in Fig. 1. Of interest are the variations which accompanied changes in pitch and intensity. The following generalizations are warranted: (1) The fundamental component was relatively more intense for soft than for loud tones. (2) The resonant frequency of the second resonator tended to be higher as pitch was raised. (3) The amount of energy in the high frequency components was relatively greater for loud than for soft tones. All three of the generalizations held for the tones represented in Fig. 2. Generalization (1) held for all of the tones investigated, whenever intensity was varied systematically. With two exceptions, generalization (3) also held for all tones investigated.

Average values of F_R and DF for the first three resonators involved in the production of the various "O's" studied are given in Table I. As already indicated, damping constants in db/sec. may be computed by multiplying values of DF by 27.3.

Typical data for the vowel "Ah," as produced

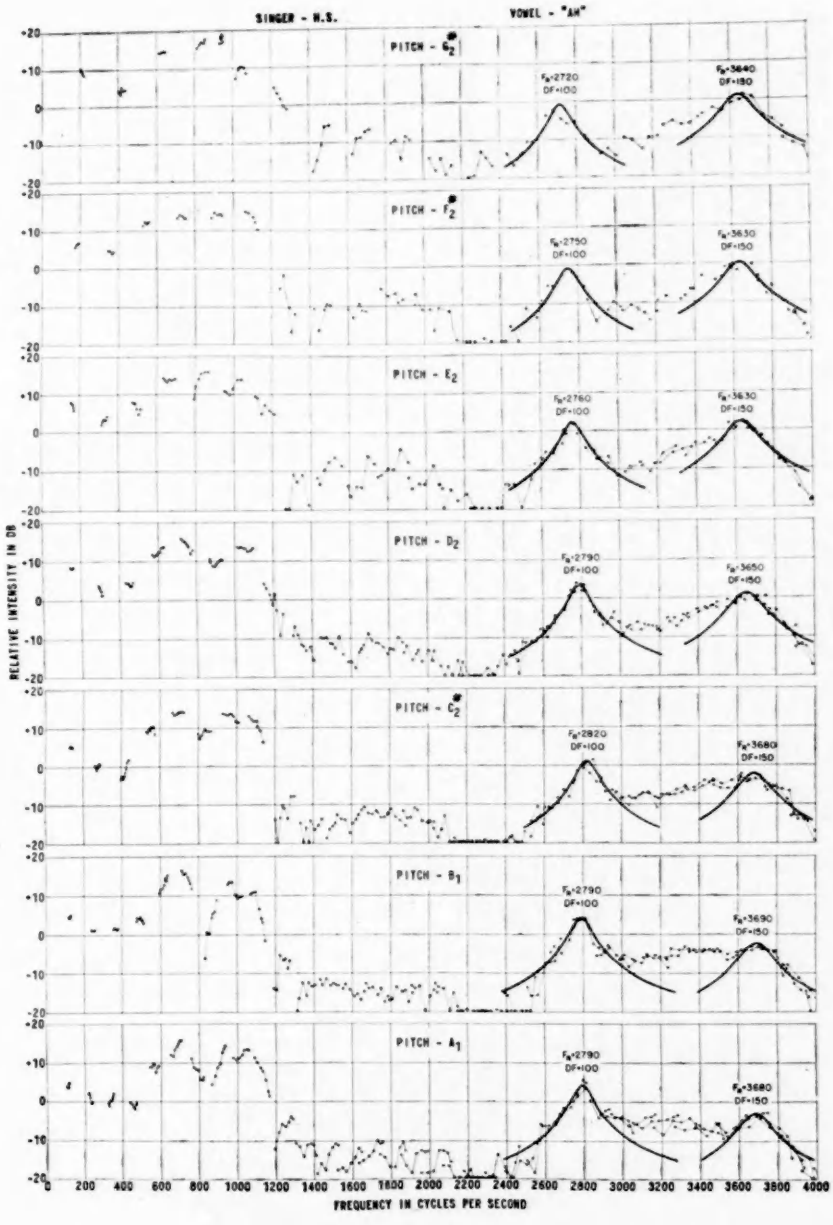

FIG. 4.

at a medium loud level by six different trained voices, are presented in Fig. 3. The pitch of each tone is indicated. The values of F_R for the first resonator range from 550 to 700 cycles, and those for the second resonator from 950 to 1160 cycles. These values are less consistent, in general, than comparable values for the vowel "O." The damping constants for "Ah" are very similar to those for "O." In fact, as a comparison of the averages in Table I will show, vowels "O" and "Ah" seem to differ significantly only in values of F_R for the first two resonators.

The method used in the investigation to determine values of F_R and DF is not always applicable. It can usually be employed readily in dealing with tones having a pitch below about $D_2^\#$ (=155.6 cycles), but it is oftentimes not applicable, except in a restricted way, to tones of a higher pitch. Fig. 4 is presented as a means of emphasizing the limitations of the method.

TABLE I. *Average values, rounded to the nearest five cycles, of F_R and DF for the first three resonators involved in the production of sustained vowels "O" and "Ah."*

	VOWEL "O"		VOWEL "Ah"	
	F_R	DF	F_R	DF
1st resonator	465	45	625	40
2nd resonator	820	50	1010	45
3rd resonator	2625	90	2680	90

Singer H.S., a trained tenor, sang the vowel "Ah" at the seven pitches indicated in the figure. He sang the seven tones consecutively, going immediately from one to the next. The order was from low to high in pitch. The tones were all essentially the same in total intensity. The analytical data were corrected for whatever intensity differences there were; so the seven graphs may be directly compared.

It will be seen that low frequency resonance curves have been omitted from Fig. 4. The principal reason for this omission was that the analytical data for the three tones represented in the three graphs at the top of the figure were such as to preclude satisfactory computations of values of F_R and DF. Reasonably satisfactory curves could have been included for the tones represented in the four bottom graphs. It was decided, however, to omit all low frequency curves to enable the reader to evaluate the data for himself. There can be little doubt that, in the case of the tones having pitches A_1, B_1, $C_2^{\#}$, and D_2, two low frequency resonators were operating whose resonant frequencies were fairly close to 700 and 1000 cycles. It might be assumed that the same two resonators, with approximately the same resonant frequencies, were also operating in the production of the tones having pitches E_2, $F_2^{\#}$, and $G_2^{\#}$. This assumption is tenable because the singer, in keeping with instructions, attempted to maintain essential constancy in the size and shape of his vocal cavities while he was singing all of the tones. Even if the assumption is open to some question, there is little to support the view that only one low frequency resonator of importance was operating in the production of the three tones of higher pitch—a view which might arise after a cursory examination of the analytical data, and especially the data for the tone of pitch $G_2^{\#}$. The authors are of the opinion that two low frequency resonators were operating in the production of all seven tones, and that the data for the three tones of highest pitch are equivocal because the harmonics were too widely separated in frequency to reveal two clear-cut regions of reinforcement in the low frequency range.

Resonance curves for high frequency resonators are included in Fig. 4. As will be seen, values of F_R for these curves do not vary to any marked extent from one tone to another. They are sufficiently inconstant, however, to indicate that there are probably no *fixed* vocal resonators of importance. On the other hand, their degree of consistency is such as to suggest the presence of relatively invariable elements in the voice of singer H.S. The voice of every singer has rather unique qualitative features, regardless of the vowel being produced; and it may very well be that individual uniqueness in voice is significantly related to the operation of one or more high frequency resonators which are relatively invariable.

The results of the investigation have shown that, in the production of sustained vowels "O" and "Ah" by trained male singers, two low frequency resonators and at least two high frequency resonators of importance are *typically* involved. A possible (though unlikely) exception to this generalization is that, in the case of high pitched tones, only one low frequency resonator is involved. The values of F_R for any one of the resonators for either vowel are somewhat variable although representative values, such as those given in Table I, may be obtained. The two low frequency resonators are undoubtedly crucial in the determination of vowel character, while the high frequency resonators probably contribute only to less basic tonal qualities. It might be said, in concluding, that there are, within certain limits, many different specific energy spectra which correspond to either "O" or "Ah," just as there are many different perceptual varieties of each vowel. "Ah" is probably more variable than "O."

Recent Experimental Investigations of Vocal Pitch in Speech

GRANT FAIRBANKS
State University of Iowa, Iowa City, Iowa
(Received January 7, 1940)

AMONG the researches which have come from the Experimental Phonetics Laboratories of the State University of Iowa, a large number have utilized acoustic measurement as a phase of experimental method. The present paper summarizes four of the most recent investigations which deal with different aspects of the same general problem: the fundamental pitch of the male voice during speech. The general purposes of the report are to illustrate experimental approaches, to present experimental findings, and to indicate directions in which future research may proceed.[1]

All of the experiments described below utilized the same technique of acoustic measurement, a modified oscillograph, designed originally by Metfessel[2] and improved successively by Simon,[3] Lewis and Tiffin,[4] and Cowan,[5] which permits measurement of fundamental sound wave frequencies from phonograph recordings by phonophotographic means. Although period-by-period measurements are possible with this instrument, the most common procedure in speech studies has been to make average measurements over consecutive intervals of 0.038 sec. each. The over-all error of this method of frequency measurement is approximately 0.5 percent, or 0.04 tone.

The first of the four experiments was concerned with the pitch characteristics of the voice during the simulation of specific emotional states.[6] Six versatile amateur male actors served as subjects, each reading the test passage, "There is no other answer. You've asked me that question a thousand times, and my reply has always been the same. It always will be the same," five times, simulating in turn contempt, anger, fear, grief and indifference. They repeated the simulations until the experimenters judged that satisfactory examples of the emotions had been secured from each subject. High quality phonograph recordings of all attempts were cut on lacquer disks.

In order to validate the simulations a method of observer-identification was used. The recordings of the 30 samples to be analyzed, six simulations of each of the five emotions, were played in random order before a group of 64 observers, advanced students of speech. Five ambiguous recordings were introduced into the random order to prevent the observers from deducing that only five different emotions were being studied. The observers were provided with lists of 12 emotional states: amusement, anger, astonishment, contempt, doubt, elation, embarrassment, fear, grief, indifference, jealousy, love. Their task was to select from the list as each recording was played the term which named the simulated emotion most accurately. The gross loudness level was kept as nearly constant as possible. The results of this judgment procedure are shown in Fig. 1, the identifications of all six simulations of each emotion being grouped. It is readily apparent that the simulations were highly satisfactory examples of the emotions to be studied.

Differences of "pitch pattern" are illustrated by the "pitch curves" of Fig. 2. For each emotion the curve presented is that of the simulation which was identified correctly by the largest percentage of observers. The abscissa is time, one-second intervals being indicated by vertical lines; the ordinate is the equal-tempered musical scale with horizontal lines marking the major triads.[7]

[1] The last three of the studies reported are from Ph.D. dissertations, completed under the direction of the author in the Department of Speech, State University of Iowa.

[2] M. Metfessel, "Technique for Objective Studies of the Vocal Art," Psychol. Monog. **36**, 1–40 (1926).

[3] C. T. Simon, "The Variability of Consecutive Wave Lengths in Vocal and Instrumental Sounds," Psychol. Monog. **36**, 41–83 (1926).

[4] D. Lewis and J. Tiffin, "A Psychophysical Study of Individual Differences in Speaking Ability," Arch. Speech **1**, 43–60 (1934).

[5] J. M. Cowan, "Pitch and Intensity Characteristics of Stage Speech," Arch. Speech **1**, suppl., 1–92 (1936).

[6] G. Fairbanks and W. Pronovost, "An Experimental Study of the Pitch Characteristics of the Voice During the Expression of Emotion," Speech Monog. **6**, 87–104 (1939).

[7] An $A=440\sim$ scale is used. In this graph and in Fig. 3 middle C at $261.6\sim$ is labeled as C_3. Now in use at the Iowa Laboratories and employed in the other graphs of this report is the new subscript system suggested by R. W. Young, *A Table Relating Frequency to Cents* (C. G. Conn Co., Elkhart, Indiana 1939), in which the zero reference frequency of $16.35\sim$ proposed by H. Fletcher,

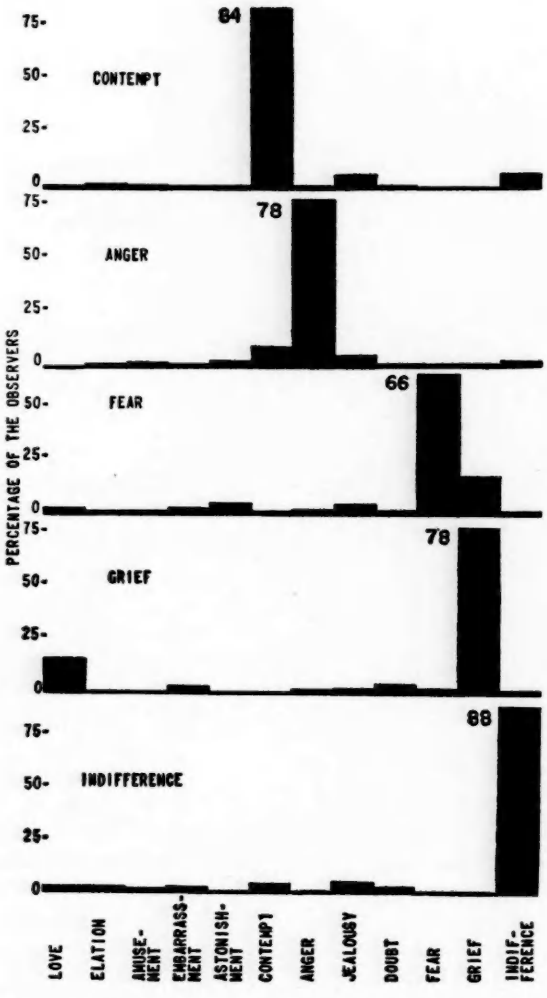

FIG. 1. Distributions of identifications of simulated emotions by observers.

Notable in this figure are the few extremely wide downward inflections in the simulation of contempt, the generally wide, rapid inflections of anger, the irregularity of the pitch changes in fear, the consistent vibrato in grief, and the lack of distinguishing features in indifference. Typical variations of pitch level are revealed in Fig. 3, which shows frequency distributions of the measured pitches for the two subjects who were generally most successful (left group) and least successful (right group) in producing identifiable simulations. The ordinate again is the musical scale; the abscissa, in each distribution, is percentage of the total measured pitches; labeled horizontal lines indicate the respective medians. It will be observed that marked differences obtain between the various emotions, both in pitch level and dispersion. For example, consider in the left group the extent of more than one octave which separates the medians for fear (225 ~) and indifference (101 ~), and the wide dispersion for contempt as compared to that for grief although the medians of the latter two simulations are only one tone apart. Viewing the group of distributions at the right of Fig. 3, it is seen that the pitch levels are by no means as clearly defined as those of the other group. This is true especially of anger and fear, and of grief and indifference. It may be suggested that these similarities contributed to the reduction in the percentage of correct identifications. Another interesting feature of this graph is that it shows both subjects to employ total pitch ranges of over three octaves in simulating the five emotions. In five of the six subjects this was found to be the case.

The most important results of the pitch measurements are summarized in Table I, in which the six subjects are considered as a group. For each measure all values for all subjects were grouped in a frequency distribution and the measure computed without reference to the individual subjects. Only in mean total pitch range, in which one value is derived from each simulation, is the number of cases six.[8] Consideration of Table I, together with the evidence of Figs. 2 and 3, will reveal the distinctive vocal pitch characteristics of five simulated emotions.

"Loudness, Pitch and Timbre of Musical Tones," J. Acous. Soc. Am. **6**, 59–69 (1934), is designated by C_0, middle C thus becoming C_4.

[8] The *median pitch level* is the median fundamental frequency. The *total pitch range* is the difference between the highest and lowest fundamental frequencies measured in a given sample and is expressed here in tones. It is computed, as are the other measures involving pitch extent, by means of the relation $N_{\text{tones}} = 19.92 \log_{10} \frac{f_1}{f_0}$, where f_1 is higher and f_0 the lower frequency. In the present studies an *inflection* is defined as a frequency modulation, either upward or downward, without interruption of phonation, while the term *shift* refers to a change in pitch which takes place between the terminal pitch of a given phonation and the initial pitch of the subsequent phonation. The *rate of pitch change* is a measure of the rapidity with which frequency is modulated per unit of time during inflections, i.e., the relative "steepness" of the inflections. For any inflection this is determined by dividing its extent in tones by its duration in seconds. A *change in direction of pitch movement* is a shift in frequency modulation from an upward direction to a downward direction, or vice versa.

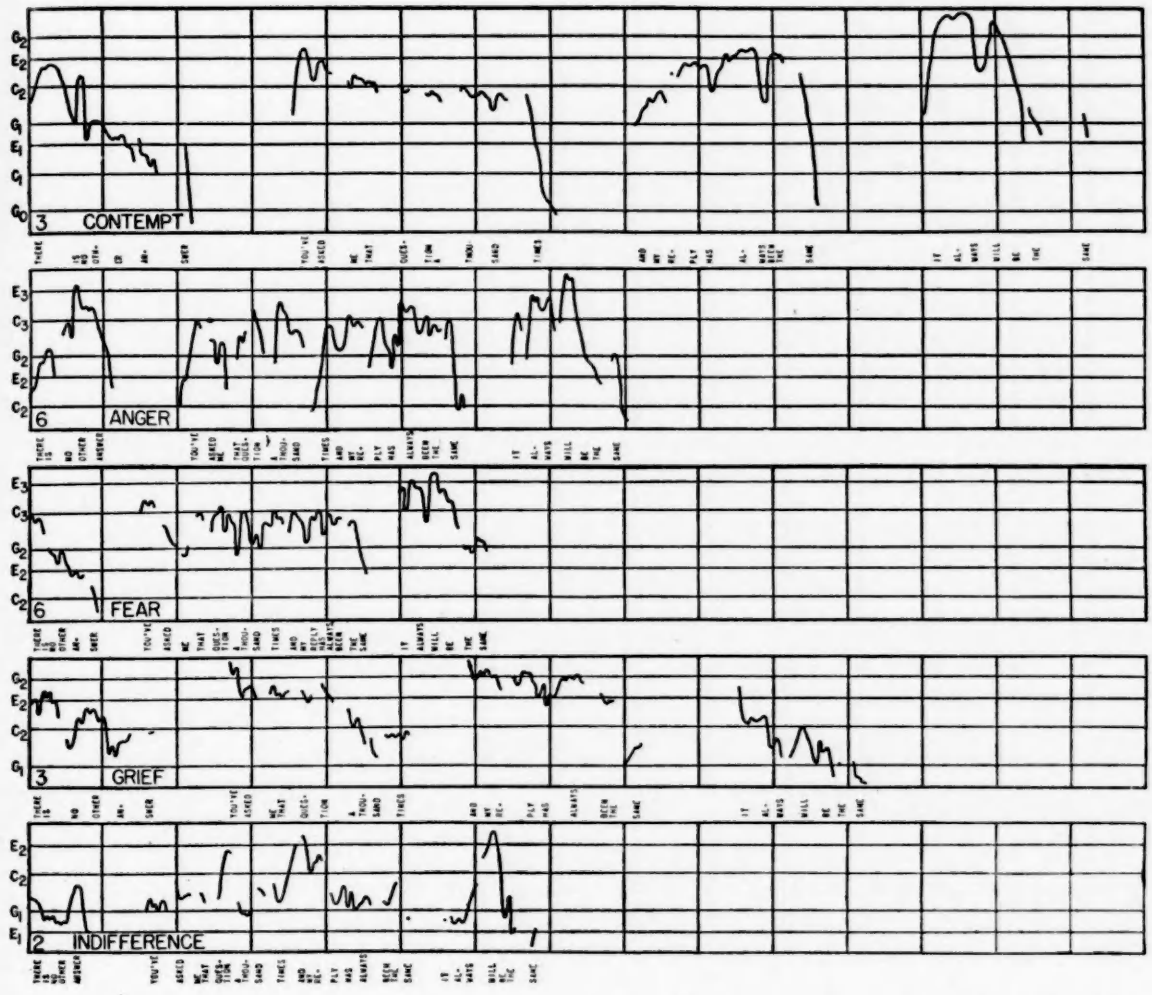

FIG. 2. Pitch curves of typical emotional simulations. For each emotion is shown the curve of the simulation which was identified correctly by the largest percentage of observers.

A second investigation[9] dealt primarily with the relationship between the fundamental vocal pitches used in speech and the range of pitches which the larynx is capable of producing, i.e., the maximum singing range. This study grew out of the clinical observation that adjustment of the pitch level is an important factor in vocal therapy, especially in such cases as hoarseness, breathiness, harshness and weak voice, in which an improper pitch level frequently operates

[9] W. Pronovost, "An Experimental Study of the Habitual and Natural Pitch Levels of Superior Speakers," Ph.D. diss., State University of Iowa, 1939.

TABLE I. *Pitch measurements of five simulated emotions.*

	CONTEMPT	ANGER	FEAR	GRIEF	INDIFFERENCE
Median pitch level (~)	124	229	254	136	108
Mean total pitch range (tones)	10.5	10.3	11.2	9.0	7.8
Mean extent of inflections (tones)	2.2	2.6	2.3	1.7	2.0
Mean extent of shifts (tones)	2.3	2.3	2.7	2.0	1.6
Mean rate of pitch change (tones/sec.)	16.8	25.6	19.0	15.6	16.6
Mean No. of changes in direction per second during phonation	8.0	8.7	9.6	10.4	8.3

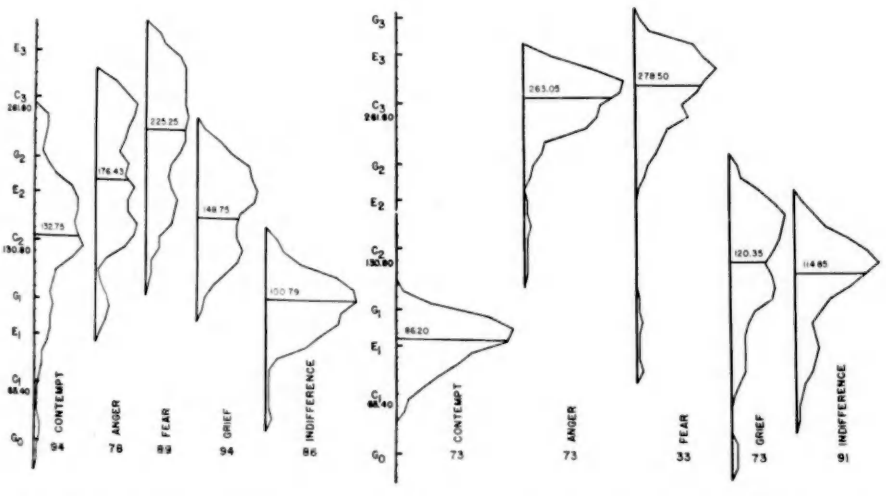

FIG. 3. Frequency distributions of pitches used in simulating the five emotions by the two subjects who were most successful (left group) and least successful (right group) in producing identifiable simulations.

causally. Superior speakers served as subjects, it being assumed that such persons approach the use of pitch levels which permit their mechanisms to function with maximum general efficiency in speech. Six such adult male speakers were selected according to the following procedure. (1) Twenty-five college students were recommended to the experimenters as having superior speech and voice usage by staff members in the Department of Speech at the State University of Iowa. (2) These students were subjected to informal but rigid scrutiny of their speaking and oral reading abilities, and rejected if they deviated from superiority in any particular. Eight students survived this procedure. (3) Each of these eight students then read the same 55-word factual prose test passage as well as he was able four times. Phonograph recordings were made. The four trials by each subject then were ranked in order of general excellence by seven trained observers and the best recording of each group of four thus was selected for further consideration. No attempt was made at this point to compare the subjects. (4) The method of paired comparisons then was used to select the best six of the eight. Each recording was paired with each other recording and played before observers who judged which of the two was superior, making possible thus a comparative ranking of the readings on the basis of general excellence.

Partial results of measurement of the recordings are shown in the frequency distributions of Fig. 4. This figure is directly comparable to Fig. 3, except that the new subscript system mentioned above has been used. The solid vertical line at the left of each distribution indicates the total normal singing range and is surmounted by a broken line showing the extension of the range into falsetto. The figure presents several interesting aspects. It is evident that these subjects employed no frequencies in the upper halves of their respective ranges, nor were any frequencies measured which were higher than the highest tones of the "normal" registers.[10] At the lower ends of the distributions it will be observed that a significant number of frequencies are used in speech which are lower than the lowest sustainable tones. This is especially true of subjects *C*, *D* and *E*. Such frequencies occur only momentarily and are troublesome in the measurement procedure since consecutive waves tend to vary markedly in period length. Extreme conservatism was used in measurement, however, all ambiguous instances being omitted, and the frequencies shown were without question present in the speech. In this regard, subject *E* presented several frequencies of the order of 25–30 ~ which

[10] Some of the measured frequencies may have been from falsetto phonations, however, since the two registers overlap by an undetermined amount at the top of normal register. Attempts were made to ascertain the lowest falsetto tones, but abandoned because of the evident unreliability.

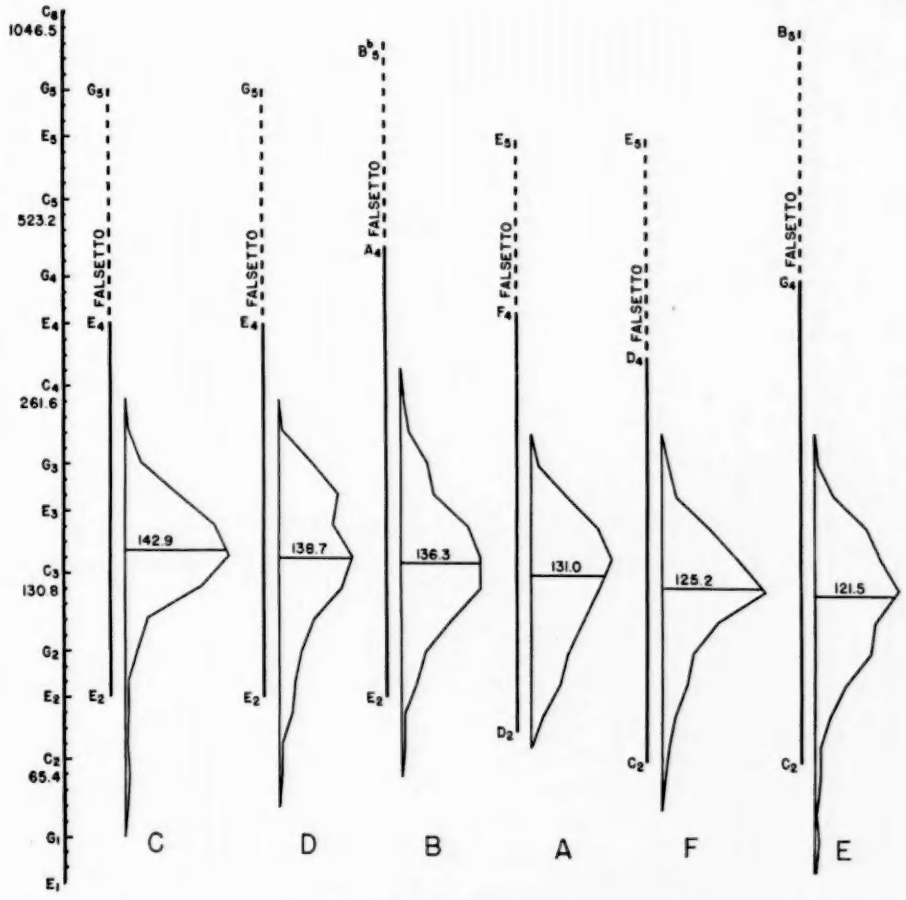

FIG. 4. Frequency distributions of pitches used by six superior speakers during the oral reading of factual prose. Solid vertical lines indicate normal singing ranges; broken lines indicate extension of the ranges into falsetto. Medians shown by the horizontal lines across the distributions.

were omitted because they could not be measured confidently, but which are readily perceptible in the phonograph recording. This phenomenon appears to be worthy of further investigation. Another remarkable feature of Fig. 4 is the proximity of the medians for the six subjects.

One object of the study was to compare various clinical devices for calculation of the "optimum" or "natural" pitch level. Without describing this procedure in detail it may be stated in summary that one method suggested by the above data was discovered to be superior in accuracy, reliability and convenience to the others which were scrutinized. This method arose from the fact that the mean ratio of the interval separating the median pitch level from the lowest sustained tone of the singing range to the total singing range including falsetto was found to be 0.25, with individual ratios varying from 0.21 to 0.28. The natural level thus is predicted 25 percent of the way up the total singing range including falsetto. Using this method, the mean deviation of the calculated natural pitch from the actual median pitch was found to be 0.45 tone, with a maximum deviation of 0.84 tone. When compared to nine other devices this method predicted the median more accurately 70 percent of the time. Its reliability was demonstrated by repeating the predictions for three of the subjects on seven consecutive days, and for one of the three subjects ten times on one day, the calculations falling within ranges of 1.5, 1.0 and 0.13 tones for the respective subjects.

The same six superior speakers also par-

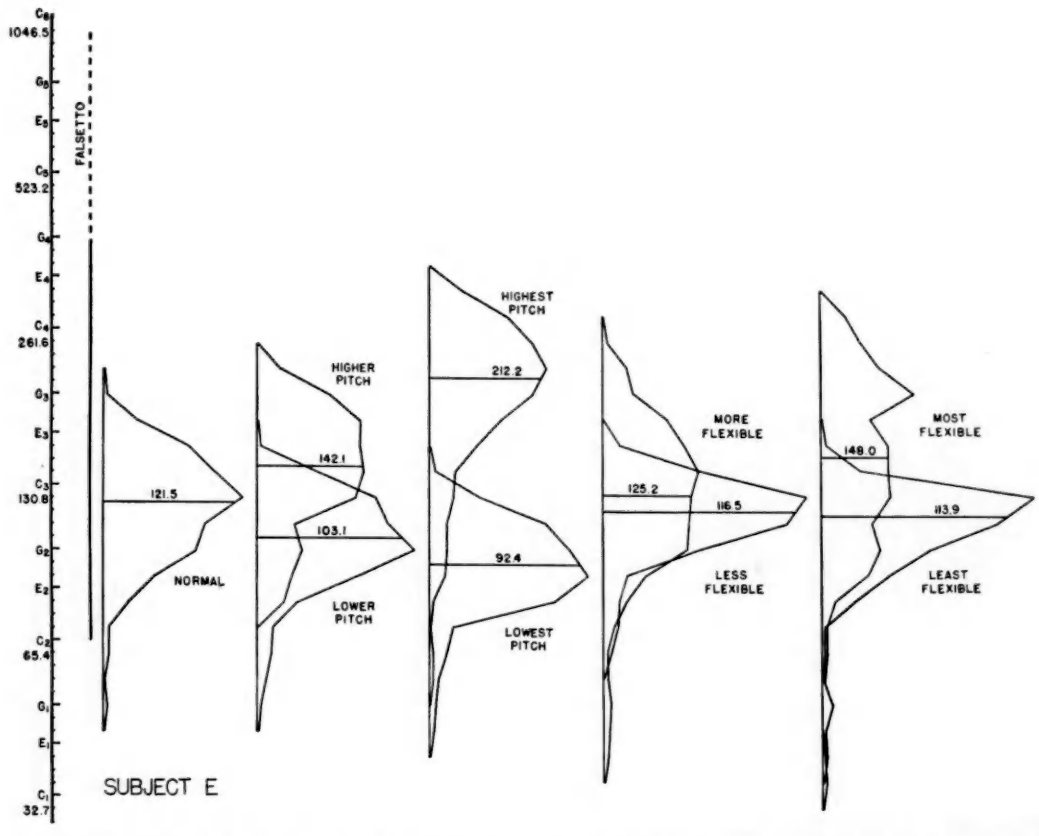

FIG. 5. Frequency distributions of pitches used by a superior speaker reading factual prose in a normal manner, and in response to instructions to reread the passage at higher and lower pitch levels and with more and less general pitch variability.

ticipated in another experiment[11] in which each subject, after listening to his recording which was measured above (hereinafter referred to as the "normal reading"), read the same passage (1) at a higher pitch level, (2) at a lower pitch level, (3) with more general variability of pitch, and (4) with less general variability of pitch. Extremes were avoided, but audition of the phonograph recordings indicated that instructions had been followed in all cases. The chief purposes of the study were to validate certain measures which had been assumed to be indicative of pitch variability and to determine whether a relationship obtains between pitch level and pitch variability.

With respect to pitch level the findings are indicated clearly in Table II, in which the group medians, i.e., medians of composite frequency distributions considering the six subjects as a group, are ranked in descending order. Expressions are given both in cycles per second and tones above the zero reference frequency, 16.35 ~. Immediately observable is the fact that an increase or decrease in variability tended to be accompanied by a systematic change in pitch level, although this should not, of course be interpreted to mean that it is impossible in a given reading to reduce the variability and raise the pitch level, or vice versa, simultaneously. The trends in pitch level may be observed graph-

TABLE II. *Composite median pitch levels.*

	CYCLES PER SECOND	TONES ABOVE 16.35 ~
Higher pitched readings	155	19.5
More variable readings	144	18.9
Normal readings	132	18.2
Less variable readings	128	17.9
Lower pitched readings	123	17.5

[11] C. W. McIntosh, Jr., "A Study of the Relationship Between Pitch Level and Pitch Variability in the Voices of Superior Speakers," Ph.D. diss., State University of Iowa, 1939.

ically in Fig. 5, which presents frequency distributions of the measured pitches for one subject. This figure also shows distributions of additional readings at still higher and lower levels and with still more and less variability. Although recorded for all subjects, these extremes were measured only in the case of this one speaker. The distributions on the first, second and fourth ordinates are those considered in this study. The additional recordings are of interest, however, since they show that the relationship between level and variability tends to prevail, at least for this subject, with even greater departures from the normal reading.

The measures which this study demonstrated to be descriptive of general pitch variability in speech are given in Table III. Here the readings are arranged from left to right in ascending order of pitch variability as indicated by the measures within the table. From the regular progression of these values it is seen generally that the higher pitched readings tend to depart from the normal readings in the direction of increased variability, while the lower pitched readings evince an opposite tendency. One reversal of this trend is noted in the mean extent of pitch shifts in the lower pitched reading, which exceeds the value for the normal reading; in two other cases adjacent values are identical. The second term in the table, *mean functional pitch range*, refers to a computation of the pitch range required to subtend the median 90 percent of the pitches used. It is employed instead of the total pitch range since the latter tends to be influenced by a few extreme frequencies (see Figs. 3, 4 and 5) and hence is not a stable measure of dispersion. The use of the median 90 percent is entirely arbitrary, but probably is a suitable fraction for descriptive purposes.

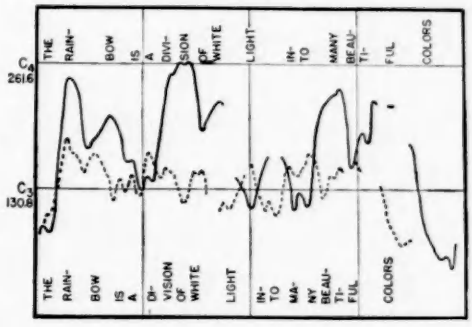
C4
261.6
C3
130.8
THE RAIN- BOW IS A DIVI- SION OF WHITE LIGHT IN- TO MANY BEAU- TI- FUL COLORS
THE RAIN- BOW IS A DI- VISION OF WHITE LIGHT IN- TO MA- NY BEAU- TI- FUL COLORS

FIG. 6. Superimposed pitch curves of the first four seconds of the more variable and less variable readings of one subject, indicated by solid and dotted lines, respectively.

It might be expected that a more rapid rate of pitch change, i.e., more abrupt inflections, would accompany increased general variability of pitch. It was interesting to discover, therefore, that the mean rate of pitch change was slower in all "instructed" readings than in the normal readings. This is explained by the fact that all instructions tended to lengthen the mean duration of inflections, as shown by another phase of the investigation not reported in this paper.[11] With the exception of the normal readings, however, the expectation that rate of pitch change is a differentiating feature of different degrees of general pitch variability is confirmed by the descending rank order shown in Table IV. It might be predicted also that changes in direction of pitch

TABLE IV.

	MEAN RATE OF PITCH CHANGE (TONES PER SECOND)
Normal reading	17.7
More variable reading	17.1
Higher pitched reading	16.2
Lower pitched reading	15.4
Less variable reading	13.2

TABLE III. *Measurements descriptive of general pitch variability. Means of performances by superior speakers reading factual prose in a normal manner, and in response to instructions to reread the passage at higher and lower pitch levels and with more and less variability. Measures arranged from left to right in ascending order of variability. All values in tones.*

	LESS VARIABLE	LOWER PITCHED	NORMAL	HIGHER PITCHED	MORE VARIABLE
Mean S.D.	1.5	1.8	1.9	2.1	2.2
Mean functional pitch range*	4.9	5.7	6.3	7.0	7.2
Mean extent of inflections	1.9	2.1	2.3	2.3	2.6
Mean extent of phonations	2.4	2.9	3.2	3.2	3.7
Mean extent of shifts	1.5	1.9	1.8	2.1	2.2

* Subtends the median 90 percent of the pitches used.

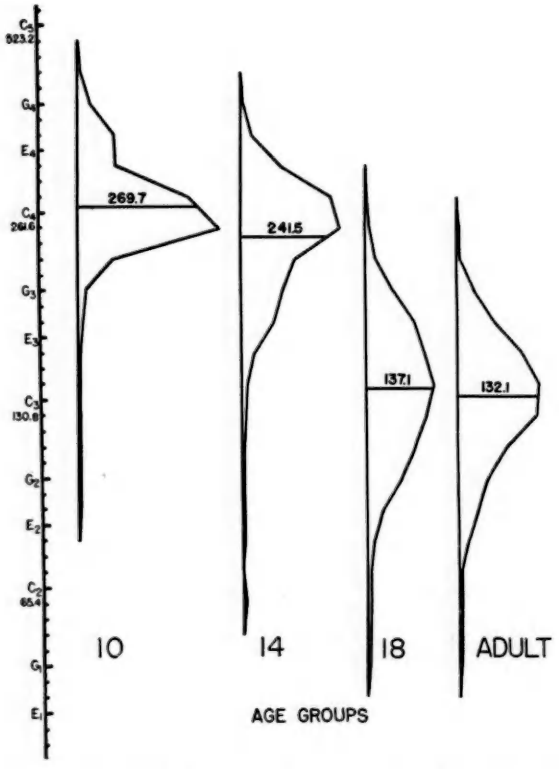

FIG. 7. Composite frequency distributions of pitches used by 10-year-old, 14-year-old, 18-year-old and adult males reading factual prose.

would become more frequent as general variability is increased. The reverse was found to be true, however. The number of changes in direction of pitch per second was larger for the less variable than for the more variable readings in all computations, considering changes during phonation only, during the total speaking time, changes one semi-tone and greater in extent, or less than one semi-tone in extent. Fig. 6, showing superimposed pitch curves of the first four seconds of the more variable and less variable readings of one subject, portrays graphically some of the features discussed above.

The fourth study[12] departed from consideration of the adult voice to make an exploratory investigation of the phenomenon of voice change in the adolescent male. Three groups of six subjects each were selected at the following age levels: (1) pre-adolescent (10 years old), (2) adolescent (14 years old), (3) post-adolescent (18 years old). In addition, the six superior adult speakers (20–30 years old) who served as subjects in the two experiments reviewed immediately above were available for direct comparison, since the same reading passage was used. The three picked groups were selected on the basis of chronological age, height, weight, intelligence, reading comprehension and speaking ability to approximate as closely as possible in all factors the medians for their respective ages. General aspects of pitch usage at the age levels studied are shown in the composite frequency distributions of Fig. 7, graphed from measurements of the oral reading performances. One feature of the graph is the fact that the median pitch levels of the 10- and 14-year-old groups approach each other in the neighborhood of C_4 while those of the 18-year-old and adult groups coincide closely near C_3, with a major difference of approximately five tones between the 14- and 18-year-old pitch levels. Apparently the most marked drop in vocal pitch during adolescence occurs during the latter four year period, and it is indicated that research in that age range would be fruitful.

Consideration of the so-called adolescent "voice breaks" formed another phase of the investigation. Oscillograms showed their most remarkable pitch characteristic to be the rapidity of the change in fundamental frequency. This change was found to be essentially different from the inflectional type of frequency modulation in

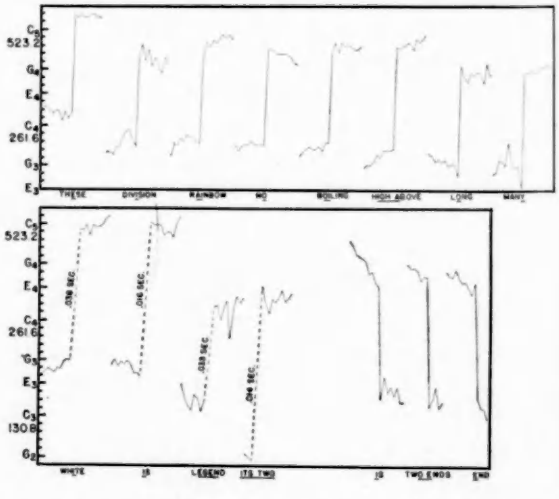

FIG. 8. Pitch changes during adolescent "voice breaks" of 15-year-old male subject Z. The abscissae are wave number.

[12] E. T. Curry, "An Objective Study of the Pitch Characteristics of the Adolescent Male Voice," Ph.D. diss., State University of Iowa, 1939.

which the period changes comparatively gradually, being instead an abrupt change from wave to wave. Twenty-five measurable breaks were found in the performances of the 14-year-old subjects and 20 at the 10-year level. At both ages a few were found which presented such an aperiodic, chaotic state that they could not be measured with certainty, although their terminal pitches were clearly identifiable. No breaks were found in the voices of the 18-year-old or adult subjects.[13] Fig. 8 presents plots of voice breaks from measurements of the periods of individual sound waves, the abscissae being wave number. Whenever possible the curves include ten waves both before and after each break, since inspection of the oscillograms showed that this number gave an adequate indication of the terminal frequencies. The dotted lines in the curves of the lower row are examples of the chaotic periods mentioned above, with the respective durations indicated. Although the curves of this figure are from an abnormal case of voice change (subject *Z*, mentioned briefly below), and are for the most part higher on the frequency scale than those at the 10- and 14-year-levels, they represent very satisfactorily the typical form of the breaks.

Measurement of the 45 instances found in the speech of the 10- and 14-year-old subjects revealed, first, that downward breaks were more frequent than expected, although outnumbered by upward breaks, and that no important or consistent differences obtained between the breaks at 10 years and those at 14 years. A second finding was that the median and also the mode extents of the breaks were close to one octave, although individual extents ranged from four to 11 tones. It was discovered further that the downward and upward breaks simply reversed themselves over approximately the same range in the pitch scale. Downward breaks occurred from frequencies close to the median speaking pitch levels for the 10- and 14-year-old subjects down to the region of the medians for the 18-year-old and adult subjects, while upward breaks occurred from the latter region to the former. In no subject was a break discovered above the individual's median; all were up from or down to the adult male pitch level which probably is to be established in a few years for these subjects, and which adjustment may be assumed from the data to be in progress at age 14. In this regard, the large number of breaks found at 10 years indicates that studies of voice during the early years also

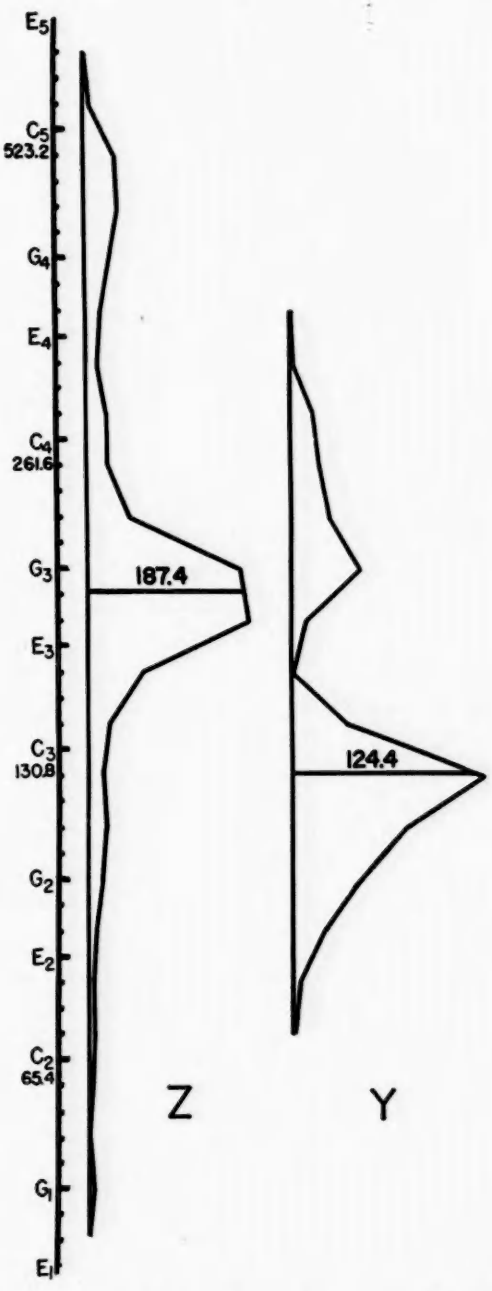

FIG. 9. Frequency distributions of pitches used during the reading of factual prose by two abnormal cases of voice change, 15-year-old male subject *Z* and 19-year-old male subject *Y*.

[13] Such breaks sometimes occur in adult voices, however.

may yield important information. An additional finding of interest is that breaks in a given direction, upward or downward, occur from frequencies within a relatively narrow range. The 11 upward breaks of the 10-year-old subjects, for example, all occur from frequencies within a range of three tones.

In addition to the groups discussed above, two atypical cases of voice change were studied. Frequency distributions of the pitches used by these subjects are shown in Fig. 9. Subject *Z* was a 15-year-old male whose voice breaks were more frequent and noticeable than in average speech at that age; subject *Y* was a 19-year-old male referred to the Speech Clinic because of the lengthy persistence of his period of voice change. The latter subject shows a clear-cut picture of partial establishment of an adult pitch level (Median = 124~) with breaks up to and down from the childhood level. His distribution in Fig. 9 is clearly bi-modal, the secondary peak indicating the upper terminal pitches of his voice breaks. In fact, he used two discrete ranges in the sample of speech measured. He differs from the 10- and 14-year-old subjects in that his most frequently used pitch level (and also his median) is approximately one octave lower than theirs, his breaks thus occurring above rather than below his median level. Fifteen-year-old subject *Z* presents a different situation. The median ($F\sharp_3$) and mode pitch levels shown in Fig. 9 are almost exactly midway between the typical childhood (C_4) and adult (C_3) levels with pitches distributed above and below this point over the extremely wide range of 3.5 octaves. As can be seen in Fig. 8, this subject's voice breaks were by no means as definitely localized as the others studied. Chaotic breaks also were more frequent. Further study may reveal that the differences shown by these two subjects are typical of certain stages of adolescent voice change.

Each of these four investigations of vocal pitch is immediately suggestive of further researches, some of which are under way at present. Among these are measurements of different types of performances, such as impromptu speaking and oral reading of poetry, comparison of inferior and superior performers, genetic studies of children's voices, and descriptive investigations of female pitch.

The Acoustical Impedance of an Infinite Hyperbolic Horn

J. E. FREEHAFER*
Massachusetts Institute of Technology, Cambridge, Massachusetts
(Received December 11, 1939)

In discussing the acoustical properties of horns, it is in general a mathematical necessity to assume plane waves. In the case of a horn in the form of an hyperboloid of one sheet, however, it is possible to avoid this assumption and to obtain an exact solution to the problem. The analysis, carried through with the aid of the differential analyzer, leads to curves representing the acoustical resistance and reactance as functions of the ratio of the radius of the throat to the wave-length. Comparison with the conical horn shows that the hyperbolic is superior.

1. INTRODUCTION

SINCE the wave equation is separable in oblate spheroidal coordinates, it is possible to investigate theoretically the performance of a horn generated by rotating an hyperbola about its conjugate axis.

The relationship between the oblate spheroidal coordinates μ, θ, and ϕ of a point and its Cartesian coordinates ξ, η, and ζ is given by the equations

$$\xi=\frac{\alpha}{2}\cosh\mu\sin\theta\cos\phi,$$

$$\eta=\frac{\alpha}{2}\cosh\mu\sin\theta\sin\phi,$$

$$\zeta=\frac{\alpha}{2}\sinh\mu\cos\theta.$$

The surface for which θ has a constant value θ_0 is an hyperboloid of revolution of one sheet whose trace on the $\xi\zeta$-plane is the hyperbola

$$\frac{\xi^2}{(\alpha/2)^2\sin^2\theta_0}-\frac{\zeta^2}{(\alpha/2)^2\cos^2\theta_0}=1.$$

It is the portion of this surface lying above the $\xi\eta$-plane which forms the hyperbolic horn. As is evident from the equation of the trace, the ζ-axis is the axis of the horn; θ_0 is the angular opening of the horn; and $(\alpha/2)\sin\theta_0$ is the radius a of the throat of the horn (see Fig. 1).

Proceeding in the usual manner, one assumes that the velocity potential $V(\mu,\theta,\phi,t)$ is given by

$$V=\psi(\mu\theta\phi)e^{i\omega t},$$

where $\omega/2\pi$ is the frequency of the wave. The differential equation for ψ (obtained by substituting V into the wave equation) is

$$\nabla^2\psi+k^2\psi=0, \tag{1}$$

where $k=2\pi/\lambda$. λ is the wave-length. Eq. (1) in oblate spheroidal coordinates is

$$\frac{1}{\cosh\mu}\frac{\partial}{\partial\mu}\left(\cosh\mu\frac{\partial\psi}{\partial\mu}\right)+\frac{1}{\sin\theta}\frac{\partial}{\partial\theta}\left(\sin\theta\frac{\partial\psi}{\partial\theta}\right)+\frac{\cosh^2\mu-\sin^2\theta}{\cosh^2\mu\sin^2\theta}\frac{\partial^2\psi}{\partial\phi^2}+\frac{k^2\alpha^2(\sinh^2\mu+\cos^2\theta)}{4}\psi=0. \tag{1a}$$

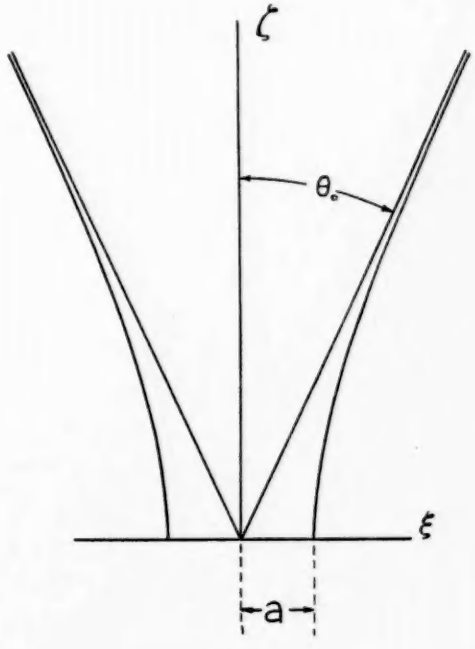

FIG. 1.

* Now at Western Reserve University, Cleveland, Ohio.

Put $\psi = F(\mu)W(\theta)\Phi(\phi)$. Eq. (1a) separates in a straightforward way, yielding the three ordinary equations

$$\frac{1}{\Phi}\frac{d^2\Phi}{d\phi^2} = -m^2 \tag{2}$$

$$\frac{1}{W\sin\theta}\frac{d}{d\theta}\left(\sin\theta\frac{dW}{d\theta}\right) - \frac{m^2}{\sin^2\theta} + \frac{\alpha^2k^2}{4}\cos^2\theta = -b, \tag{3}$$

$$\frac{1}{F\cosh\mu}\frac{d}{d\mu}\left(\cosh\mu\frac{dF}{d\mu}\right) + \frac{m^2}{\cosh^2\mu} + \frac{\alpha^2k^2}{4}\sinh^2\mu = b, \tag{4}$$

where m^2 and b are the separation constants. Since it is desired to treat the important special case of cylindrical symmetry about the axis of the horn, m^2 is taken equal to zero. Because ψ must be single-valued, Φ is simply a constant and may be taken equal to unity. A change in the independent variables from θ to $z=\cos\theta$ and from μ to $x=\sinh\mu$ converts (3) to

$$\frac{d}{dz}\left[(1-z^2)\frac{dW}{dz}\right] + (b - c^2z^2)W = 0 \tag{3a}$$

and (4) to

$$\frac{d}{dx}\left[(1+x^2)\frac{dF}{dx}\right] - (b + c^2x^2)F = 0, \tag{4a}$$

where $c^2 = -k^2\alpha^2/4$.

Since in the region interior to the horn $0 \le \theta \le \theta_0$, it follows that $z_0 \le z \le 1$ where $z_0 = \cos\theta_0$. Eq. (3a) has a regular singular point at $z=1$. One integral of the equation is analytic and the other independent integrals have logarithmic singularities at this point. In order that ψ may be finite along the axis of the horn where $z=1$, the analytic integral must be used. The condition that $\nabla\psi$ have no component normal to the boundary at the surface of the horn requires $W'(z_0)=0$, and limits b in the usual way to a set of characteristic values which are functions of c^2. Let b_n denote the sequence of characteristic values such that $b_n < b_{n+1}$. W_n will be used to denote the function whose value is unity at $z=1$ and which satisfies the differential equation

$$\frac{d}{dz}\left[(1-z^2)\frac{dW_n}{dz}\right] + (b_n - c^2z^2)W_n = 0. \tag{3b}$$

Then $W_n(1)=1$ and $W_n'(z_0)=0$. These characteristic functions are to be examined more fully later in the discussion.

To each characteristic value b_n, there corresponds a radial function F_n which satisfies the equation

$$\frac{d}{dx}\left[(1+x^2)\frac{dF_n}{dx}\right] - (b_n + c^2x^2)F_n = 0 \tag{4b}$$

and which when multiplied by $e^{i\omega t}$ must represent for large values of x a wave traveling in the direction of increasing x.

Any function ψ satisfying the boundary conditions along the surface of the horn and at infinity may be represented by

$$\psi(\mu, \theta) = \sum_{n=0}^{\infty} A_n W_n(\cos\theta) F_n(\sinh\mu).$$

The A_n's are constants which depend upon conditions at the throat of the horn. Let the horn be driven by a plane piston coincident with the surface $\mu=0$ and oscillating at a constant frequency $\omega/2\pi$. The speed u of the air at the piston is then given by $u = u_0e^{i\omega t}$, where u_0 is a constant. The velocity must be perpendicular to the piston, and hence $-u$ must equal the component of the gradient of the velocity potential in the direction of increasing μ when $\mu=0$.

$$-u = \nabla_\mu\psi e^{i\omega t}\Big|_{\mu=0} = e^{i\omega t}(2/\alpha\cos\theta)\frac{\partial\psi}{\partial\mu}\Big|_{\mu=0},$$

$$-u_0 = (2/\alpha\cos\theta)\sum_{n=0}^{\infty} A_n W_n(z) F_n'(0),$$

$$-u_0\alpha z/2 = \sum_{n=0}^{\infty} A_n W_n(z) F_n'(0).$$

Multiplication by $W_m(z)$ and integration in respect to z from z_0 to 1 yields

$$(-u_0\alpha/2)\int_{z_0}^{1} W_m(z)z\,dz = \sum_{n=0}^{\infty} A_n F_n'(0)\int_{z_0}^{1} W_n W_m\,dz.$$

Now because of the orthogonal properties of the

W's which may be proved in the usual way from Eq. (3b) and the boundary condition that $W_n'(z_0)=0$, it is true that

$$\int_{z_0}^{1} W_n W_m dz = 0 \quad \text{if} \quad n \neq m.$$

Therefore

$$(-u_0\alpha/2)\int_{z_0}^{1} W_m(z) z dz = A_m F_m'(0)\int_{z_0}^{1} W_m^2 dz,$$

$$A_m = \left(-u_0\alpha\int_{z_0}^{1} W_m z dz\right) \Big/ 2F_m'(0)\int_{z_0}^{1} W_m^2 dz. \quad (5)$$

The excess pressure p due to the wave at any point in the sound field is given by $p=\rho_0 \partial V/\partial t$ where ρ_0 is the density of the undisturbed medium. The pressure p_0 at the piston is therefore

$$p_0 = \rho_0 \psi(0,\theta)\frac{d}{dt}e^{i\omega t}$$

$$= \rho_0 i\omega e^{i\omega t}\sum_{n=0}^{\infty} A_n W_n(\cos\theta) F_n(0).$$

To obtain the total force H exerted on the piston by the wave, it is necessary to integrate over the face of the piston. Let dr be the width of an annular ring of radius r about the center. Because of symmetry,

$$H = 2\pi\int_0^a p_0 r dr.$$

But $r^2=\xi^2(0,\theta\phi)+\eta^2(0,\theta\phi)=(\alpha \sin\theta/2)^2$ and $2rdr = \alpha^2/2 \sin\theta\cos\theta d\theta$. In terms of $z=\cos\theta$, $2rdr = -\alpha^2/2zdz$. Hence

$$H = (-\pi\alpha^2/2)\int_1^{z_0} p_0 z dz = (\pi\alpha^2/2)\int_{z_0}^{1} p_0 z dz$$

$$= (\pi\alpha^2/2)\rho_0 i\omega e^{i\omega t}\sum_{n=0}^{\infty} A_n F_n(0)\int_{z_0}^{1} W_n z dz.$$

Using Eq. (5), one obtains

$$H = -\frac{\pi\alpha^3}{4}\rho_0 i\omega(u_0 e^{i\omega t})\sum_{n=0}^{\infty}\frac{F_n(0)}{F_n'(0)}\frac{\left[\int_{z_0}^{1} W_n z dz\right]^2}{\int_{z_0}^{1} W_n^2 dz}.$$

By definition, the acoustic impedance Z of the horn is given by

$$Z \equiv H/u = (-\pi\alpha^3/4)\rho_0 i\omega \times \sum_{n=0}^{\infty}\frac{F_n(0)}{F_n'(0)}\frac{\left[\int_{z_0}^{1} W_n z dz\right]^2}{\int_{z_0}^{1} W_n^2 dz}. \quad (6)$$

It is necessary before Z may be evaluated to obtain the W's and F's.

2. The Characteristic Functions

The first step in obtaining the W's is to find the characteristic values. The b's are functions of c^2 and the W's are functions of both z and c^2. Let it be assumed that

$$b = \sum_{m=0}^{\infty} e_m c^{2m} \quad \text{and} \quad W = \sum_{s=0}^{\infty} f_s c^{2s},$$

where the e's are constants and the f's are functions of z. Substituting these expressions in Eq. (3a) and equating the coefficients of the powers of c^2 in the resulting series to zero, one obtains the condition

$$\frac{d}{dz}\left[(1-z^2)\frac{df_s}{dz}\right] + e_0 f_s = (z^2-e_1)f_{s-1} - \sum_{m=2}^{s} e_m f_{s-m}. \quad (7)$$

To investigate (7), use is made of the theorem that if

$$\frac{d}{dx}\frac{(rdv)}{dx} + sv = 0,$$

then an integral of the equation

$$\frac{d}{dx}\frac{(rdy)}{dx} + sy = p$$

is given by $\quad y = v\int \frac{\int pv dx}{rv^2} dx.$

Now f_0 satisfies the equation

$$\frac{d}{dz}\left[(1-z^2)\frac{df_0}{dz}\right] + e_0 f_0 = 0$$

and therefore

$$f_s = f_0\int \frac{\int\left[(z^2-e_1)f_{s-1} - \sum_{m=2}^{s} e_m f_{s-m}\right] f_0 dz}{(1-z^2)f_0^2} dz.$$

If W is to be analytic at $z=1$ and $W'(z_0)=0$, it is necessary that $f_s'(1)$ be finite and $f_s'(z_0)=0$. This requires that

$$\int[(z^2-e_1)f_{s-1}-\sum_{m=2}^{s}e_m f_{s-m}]f_0 dz=0,$$

when $z=z_0$ or 1. The first condition is met by taking z_0 as the lower limit of integration; the second, by choosing e_s so that

$$\int_{z_0}^{1}[(z^2-e_1)f_{s-1}-\sum_{m=2}^{s}e_m f_{s-m}]f_0 dz=0 \qquad (8)$$

or

$$e_s=\frac{\int_{z_0}^{1}[(z^2-e_1)f_{s-1}-\sum_{m=2}^{s-1}e_m f_{s-m}]f_0 dz}{\int_{z_0}^{1}f_0^2 dz}. \qquad (9)$$

Eq. (9) does not yield e_1. Eq. (8) for $s=1$ is

$$\int_{z_0}^{1}(z^2-e_1)f_0^2 dz=0$$

and hence

$$e_1=\int_{z_0}^{1}z^2 f_0^2 dz\Big/\int_{z_0}^{1}f_0^2 dz.$$

e_s may evidently be computed if we know $e_{s-1}, e_{s-2}\cdots e_0$ and $f_{s-1}, f_{s-2}\cdots f_0$. Then f_s can be determined from the expression

$$f_s=f_0\int_1^z\frac{\int_{z_0}^{z}[(z^2-e_1)f_{s-1}-\sum_{m=2}^{s}e_m f_{s-m}]f_0 dz}{(1-z^2)f_0^2}dz,$$

which by the use of (8) becomes

$$f_s=f_0\int_1^z\frac{\int_{1}^{z}[(z^2-e_1)f_{s-1}-\sum_{m=2}^{s}e_m f_{s-m}]f_0 dz}{(1-z^2)f_0^2}dz.$$

The lower limit of the outer integral is chosen so that $f_s(1)=0$ for $s\neq 0$ and hence $W(1)$ is independent of c^2.

Before the process outlined above can be started, it is necessary to know f_0. Since f_0 satisfies the Legendre equation

$$\frac{d}{dz}\left[(1-z^2)\frac{df_0}{dz}\right]+e_0 f_0=0$$

and is finite at $+1$, it follows that $f_0=P_n(z)$ where $n(n+1)=e_0$. n must be chosen so that $P_n'(z_0)=0$. Roots of the equation $P_n'(z_0)=0$ may be calculated by the method of Bholanath Pal.[1] The first few roots for $z_0=\cos 15°$ and $\cos 30°$, are given below.

$z_0=\cos 15°$		$z_0=\cos 30°$	
n	e_0	n	e_0
0.00	0.0	0.00	0.0
14.12	213.5	6.83	53.5
26.29	717.5	12.89	179.0
38.35	1509.	18.93	358.5

For each of the infinite number of roots, one obtains a value of e_0 and, by carrying out the process just outlined, obtains a characteristic value of b and the corresponding characteristic function.

The first characteristic value and function are readily calculated from the knowledge that $P_0(z)=1$ and has a zero derivative everywhere. Thus for all values of z_0, $e_0=0$ and $f_0=1$. These values lead to the following expressions for the first few e's and f's:

$$e_0=0,$$

$$e_1=\frac{z_0^2+z_0+1}{3},$$

$$e_2=\frac{(1-z_0)^3}{36}\left[\frac{5z_0^3-12z_0-8}{15}-\frac{z_0^2(z_0+1)^2}{2}\sum_{n=4}^{\infty}\frac{1}{n}\left(\frac{1-z_0}{2}\right)^{n-4}\right],$$

$$f_0=1,\quad f_1=(1-z)\left(\frac{1+z}{6}-\frac{1-3e_1}{3(1-z)}\ln\frac{1+z}{2}\right). \qquad (10)$$

It is not practicable by this method to calculate the second and higher characteristic values. A somewhat different method of approach must be used. Let p_m stand for the set of characteristic functions for which $c^2=0$, and d_m be the corresponding set of characteristic values. Then $p_m(1)$ is finite, $p_m'(z_0)=0$, and

$$\frac{d}{dz}[(1-z^2)p_m']+d_m p_m=0. \qquad (11)$$

[1] B. Pal, Bulletin of the Calcutta Math. Soc. **9**, 85; (1917) **10**, 187 (1918).

The p's are, of course, Legendre functions of the first kind of appropriate order. Because the p's form an orthogonal and complete set, it is possible to expand the f's in the form

$$f_s = \sum_{m=0}^{\infty} a_{sm} p_m, \tag{12}$$

where

$$a_{sm} = \int_{z_0}^{1} f_s p_m dz \Big/ \int_{z_0}^{1} p_m^2 dz.$$

It is also possible to expand $z^2 f_s$ in a series of p's.

$$z^2 f_s = \sum_{m=0}^{\infty} c_{sm} p_m, \tag{13}$$

where

$$c_{sm} = \int_{z_0}^{1} z^2 f_s p_m dz \Big/ \int_{z_0}^{1} p_m^2 dz.$$

Replacing f_s by its series expansion, one obtains

$$c_{sm} = \sum_{n=0}^{\infty} a_{sn} k_{mn},$$

where

$$k_{mn} = \int_{z_0}^{1} z^2 p_m p_n dz \Big/ \int_{z_0}^{1} p_m^2 dz.$$

Substitution of (12) and (13) in (7), yields

$$\sum_{m=0}^{\infty} \left\{ \frac{d}{dz}[(1-z^2)p_m'] + e_0 p_m \right\} a_{sm} = \sum_{m=0}^{\infty} c_{s-1,m} p_m - \sum_{t=1}^{s} e_t \sum_{m=0}^{\infty} a_{s-t,m} p_m.$$

It must be noted that f_0 is some one of the p's. As a matter of fact f_0 for the $(r+1)^{st}$ characteristic function is p_r and likewise e_0 is d_r. Using this fact together with (11), one obtains

$$\sum_{m=0}^{\infty} (d_r - d_m) a_{sm} p_m = \sum_{m=0}^{\infty} c_{s-1,m} p_m - \sum_{m=0}^{\infty} \left(\sum a_{s-t,m} e_t\right) p_m.$$

By equating coefficients of p_m on each side of the equation, one is led to

$$a_{sm} = \frac{c_{s-1,m} - \sum_{t=1}^{s} a_{s-t,m} e_t}{d_r - d_m} \quad \text{when } m \neq r.$$

To avoid an infinite value of a_{sr}, it is necessary that

$$c_{s-1,r} = \sum_{t=1}^{s} a_{s-t,r} e_t$$

or

$$e_s = \frac{c_{s-1,r} - \sum_{t=1}^{s-1} a_{s-t,r} e_t}{a_{0r}},$$

but

$$a_{0r} = \int_{z_0}^{1} f_0 p_r dz \Big/ \int_{z_0}^{1} p_r^2 dz.$$

Since $f_0 = p_r$, $a_{0r} = 1$. Thus the condition from which e_s may be computed is

$$e_s = c_{s-1,r} - \sum_{t=1}^{s-1} a_{s-t,r} e_t.$$

Since

$$c_{s-1,r} = \sum_{n=0}^{\infty} a_{s-1,n} k_{rn},$$

$$e_s = \sum_{n=0}^{\infty} a_{s-1,n} k_{rn} - \sum_{t=1}^{s-1} a_{s-t,r} e_t. \tag{14}$$

Note that

$$e_1 = \sum_{n=0}^{\infty} a_{0n} k_{rn} = k_{rr}$$

since $a_{0n} = 0$ when $n \neq r$. Finally

$$a_{sm} = \frac{\sum_{n=0}^{\infty} a_{s-1,n} k_{mn} - \sum_{t=1}^{s} a_{s-t,m} e_t}{d_r - d_m}. \tag{15}$$

a_{sr} is arbitrary insofar as the above analysis is concerned. It is convenient, however, to choose its value so that $f_s(1) = 0$ when $s \neq 0$.

By means of (14) and (15), it is possible to determine the e's and a's in turn. From Pal's investigation of the roots of $P_n'(z_0) = 0$, it follows that for a 15° horn (i.e., $z_0 = \cos 15°$), the sequence of p's is P_0, $P_{14.12}$, $P_{26.29}$, $P_{38.35}$, $\cdots$; and for the 30° horn, P_0, $P_{6.83}$, $P_{12.89}$, $P_{18.93}$, $\cdots$. The method indicated above yields:

15° Horn

$$\left.\begin{aligned} b_0 &= 0.96631c^2 - 1.607 \times 10^{-6} c^4 \cdots \\ b_1 &= 213.5 + 0.977c^2 + 0(10^{-6} c^4) \cdots \\ b_2 &= 717.5 + 0.980c^2 + 0(10^{-8} c^4) \cdots \end{aligned}\right\} \tag{16a}$$

30° Horn

$$\left.\begin{aligned} b_0 &= 0.87201c^2 - 9.0486 \times 10^{-5} c^4 \cdots \\ b_1 &= 53.5 + 0.914c^2 + 7.0 \times 10^{-5} c^4 \cdots \\ b_2 &= 179.0 + 0.919c^2 + 0(10^{-5} c^4) \cdots \end{aligned}\right\} \tag{16b}$$

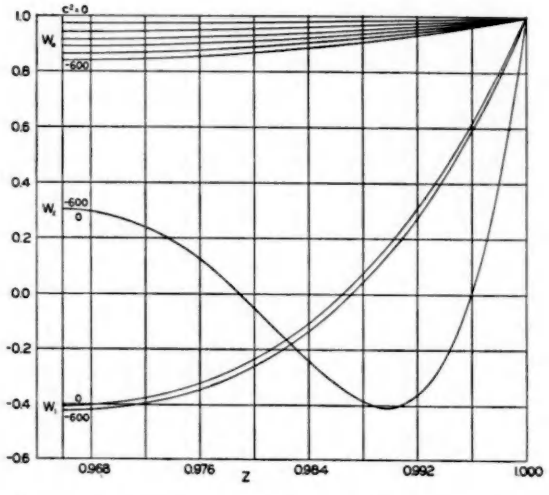

FIG. 2. Characteristic functions for the 15° horn. W_0 is given at intervals in c^2 of -100. The individual W_2 curves are not resolved.

The integrals involved in the calculation of the coefficients in the above expressions unfortunately cannot be conveniently evaluated analytically. They were accordingly obtained graphically with the aid of the tables of Hj. Tallqvist.[2]

The range of values of c^2 is obtained from the fact that waves whose lengths are shorter than the radius of the throat of the horn are not appreciably diffracted. Now $c^2=-(2\pi a/\lambda \sin\theta_0)^2$. Since a is to be the smallest value of λ, $c^2>-(2\pi/\sin\theta_0)^2$. Thus for a 15° horn $-600<c^2<0$ and for a 30° horn $-160<c^2<0$.

If no other method were available, it would be possible to evaluate the characteristic functions by using the scheme outlined above. Since the differential analyzer was available, however, it was more convenient to obtain the characteristic values from (16a) and (16b) and then to use the machine for the calculation of the functions. Figs. 2 and 3 were obtained in this manner.

The expression for Z involves

$$\left(\int_{z_0}^{1} W_n z dz\right)^2 \Big/ \int_{z_0}^{1} W_n^2 dz.$$

In the case of $n=0$, an expansion for this quantity in powers of c^2 may be obtained by using the expansion for W_0. Thus

$$\left(\int_{z_0}^{1} W_0 z dz\right)^2 \Big/ \int_{z_0}^{1} W_0^2 dz = E_0 + E_1 c^2 \cdots$$

where

$$E_0 = \left(\int_{z_0}^{1} z dz\right)^2 \Big/ \int_{z_0}^{1} dz$$

and

$$E_1 = \frac{2\int_{z_0}^{1} z dz}{\left(\int_{z_0}^{1} dz\right)^2}\left(\int_{z_0}^{1} dz \cdot \int_{z_0}^{1} f_1 z dz - \int_{z_0}^{1} z dz \cdot \int_{z_0}^{1} f_1 dz\right)$$

f_1 is given by (10). After considerable manipulation, it can be shown that

$$E_0 = \frac{(1-z_0)^2(1+z_0)}{4} = \frac{1}{2}\sin^2\theta_0 \cos^2\frac{\theta_0}{2},$$

$$E_1 = -\frac{(1-z_0)^4(1+z_0)^2}{24} \times\left\{\frac{1-z_0}{3}+\frac{z_0(1+z_0)}{2}\sum_{n=0}^{\infty}\frac{(1-z_0)^n}{2^n(n+4)}\right\}.$$

For the 15° horn $E_0=0.03292$ and $E_1=-5.47\times10^{-8}$; and for the 30° horn $E_0=0.1166$ and $E_1=-1.21\times10^{-5}$. Accordingly, for the values of c^2 which are of interest, it is necessary to retain only the first term in the case of the 15° horn, and the first two in the 30° case.

When $c^2=0$, W_1 equals $P_{14.12}$ and $P_{6.83}$ for the 15° and 30° horns, respectively. By graphical means, it is possible to show that

$$\frac{\left(\int_{\cos 15°}^{1} zP_{14.12}dz\right)^2}{\int_{\cos 15°}^{1} P^2_{14.12}dz} = 4\times10^{-6}$$

and

[2] H. Tallqvist, Societas Scientiarium Fennica. Commentationes Physico-Mathematicae VI 3 and VI 10 (1932). The following integrals were obtained graphically and used in the calculation of the coefficients appearing in (16a) and (16b). The upper and lower limits of the integrals below are 1 and cos 15°, respectively.

$$\int P^2_{14.12}dz=0.00555, \quad \int z^2P^2_{14.12}dz=0.00542,$$
$$\int P^2_{26.29}dz=0.00308, \quad \int z^2P^2_{26.29}dz=0.00302,$$
$$\int z^2P_{14.12}dz=3\times10^{-4}, \quad \int z^2P_{14.12}P_{26.29}dz=5\times10^{-6}.$$

The upper and lower limits of the following integrals are 1 and cos 30°, respectively.

$$\int P^2_{6.83}dz=0.0222, \quad \int z^2P^2_{6.83}dz=0.0203,$$
$$\int P^2_{12.89}dz=0.0123, \quad \int z^2P^2_{12.89}dz=0.0113,$$
$$\int z^2P_{6.83}dz=0.00388, \quad \int z^2P_{6.83}P_{12.89}dz=9.3\times10^{-4}.$$

$$\frac{\left(\int_{\cos 30^\circ}^{1} zP_{6.83}dz\right)^2}{\int_{\cos 30^\circ}^{1} P^2_{6.83}dz} = 2.2\times10^{-4}.$$

Figs. 2 and 3 show that W_1 does not change a great deal with c^2 especially in the 15° case. The change in any case is such that

$$\left(\int_{z_0}^{1} W_1 z dz\right)^2 \Big/ \int_{z_0}^{1} W_1^2 dz$$

decreases as c^2 decreases. If we may assume that $|F_1(0)/F_1'(0)|$ is of about the same magnitude as $|F_0(0)/F_0'(0)|$, we are justified in neglecting all but the first term in the expansion of Z.

3. The Radial Functions

The next step in calculating Z is to obtain $F_n(0)/F_n'(0)$. F_n satisfies Eq. (4b) and must represent a wave traveling outward from the throat of the horn. The problem which arises at this point is that of analytic continuation. Provided that c^2 and x are not both too close to zero, the WKB approximation is valid. Thus

$$F_n = \frac{\exp c\int\left(\frac{\delta_n+x^2}{1+x^2}\right)^{\frac{1}{2}}dx}{[(1+x^2)(\delta_n+x^2)]^{\frac{1}{4}}} \quad \text{where} \quad \delta_n = b_n/c^2.$$

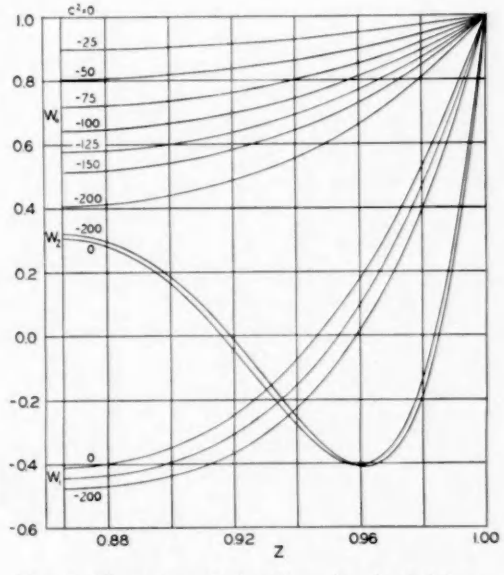

Fig. 3. Characteristic functions for the 30° horn.

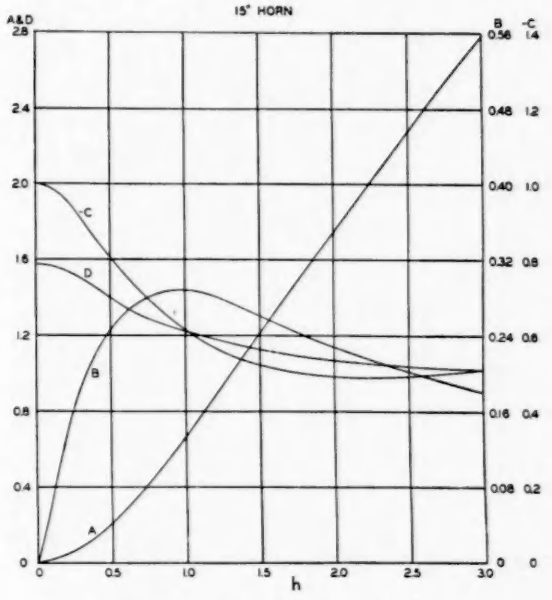

Fig. 4. Joining factors for the 15° case.

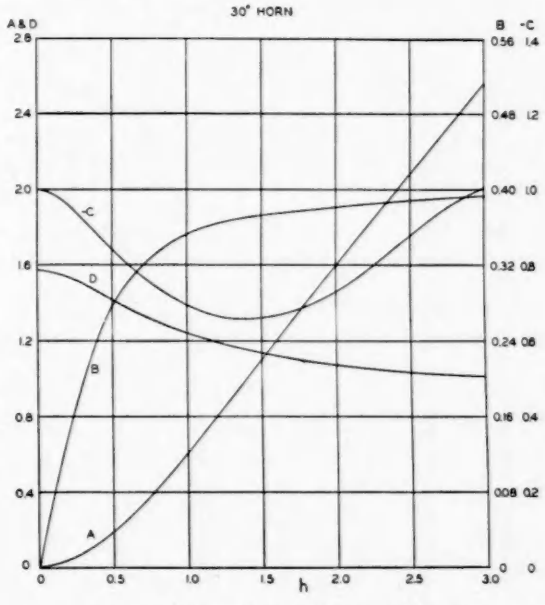

Fig. 5. Joining factors for the 30° case.

Now δ_0 is nearly equal to unity; and if x^2 is large compared with one, very little error is introduced by putting $\delta_0 = 1$.

$$F_0 = \frac{e^{cz}}{(1+x^2)^{\frac{1}{4}}} = \frac{e^{-ihz}}{(1+x^2)^{\frac{1}{4}}}, \quad \text{where } h = (-c^2)^{\frac{1}{2}}. \quad (17)$$

h is the positive square root. Let q_1 and q_2 be integrals of (4b) which satisfy the conditions:

$q_1(0)=0$, $q_1'(0)=1$; and $q_2(0)=1$, $q_2'(0)=0$. In the region in which (17) is valid,

$$Aq_1+Bq_2=\sin hx/(1+x^2)^{\frac{1}{4}}, \qquad (18a)$$

$$Cq_1+Dq_2=\cos hx/(1+x^2)^{\frac{1}{4}}, \qquad (18b)$$

where A, B, C and D are the joining factors. They are functions of c^2 as well as of z_0. Unfortunately the region about the origin in which power series expansions of q_1 and q_2 converge does not overlap the region in which (17) is valid. By means of the differential analyzer, however, values of q_1 and q_2 together with their derivatives may be found at a point suitable for joining. $x=5$ has been taken as a point which is both convenient and proper provided c^2 is not too nearly equal to zero. Figs. 4 and 5 give values of A, B, C and D which were obtained in this manner.

The joining factors may be calculated without using the analyzer when $c^2=0$; for then, Eq. (4b) is simply

$$\frac{d}{dx}[(1+x^2)F']=0.$$

Therefore $q_1=\tan^{-1}x$ and $q_2=1$. Since the expressions on the right of (18) merely represent the integrals asymptotically, it is proper to join them to the left-hand members only at $x=\infty$. Thus if $h=0$, (18b) becomes

$$C\tan^{-1}x+D=(1+x^2)^{-\frac{1}{4}},$$

which yields on differentiation

$$C/(1+x^2)=-x/(1+x^2)^{\frac{3}{4}}.$$

Therefore $C=-x/(1+x^2)^{\frac{1}{4}}$ provided x is large. Evidently the proper value is given by

$$C=\lim_{x\to\infty} -x/(1+x^2)^{\frac{1}{4}}=-1.$$

Likewise

$$D=\lim_{x\to\infty}(1+x^2)^{-\frac{1}{4}}+\tan^{-1}x=\pi/2.$$

In a similar way, it may be shown that when $c^2=0$, $A=0=B$.

Now

$$F_0=\frac{\cos hx-i\sin hx}{(1+x^2)^{\frac{1}{4}}}=(C-iA)q_1+(D-iB)q_2.$$

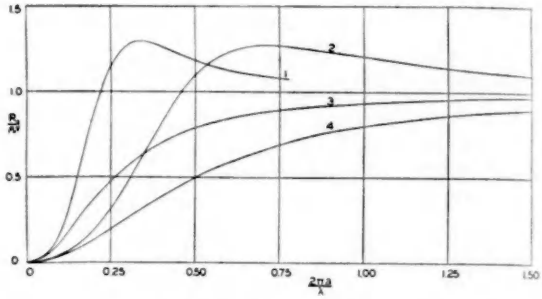

FIG. 6. Transmission coefficients. Curves 1, 2, 3 and 4, are for the 15° hyperbolic, 30° hyperbolic, 15° conical, and 30° conical horns, respectively.

Therefore

$$\frac{F_0(0)}{F_0'(0)}=G_1+iG_2,$$

where

$$G_1=\frac{AB+CD}{A^2+C^2} \quad \text{and} \quad G_2=\frac{AD-BC}{A^2+C^2}.$$

4. The Acoustic Impedance

If only the first term of the series occurring in (6) be retained and if the substitutions

$$\frac{\left(\int_{z_0}^{1}W_0 z\,dz\right)^2}{\int_{z_0}^{1}W_0^2\,dz}=\frac{1}{2}\sin^2\theta_0\cos^2\frac{\theta_0}{2}$$

$$\omega=\frac{2vh}{\alpha}$$

(where v is the velocity of sound in air) be introduced, one obtains

$$Z_0=\frac{Z}{\pi a^2}=\rho_0 vh\cos^2\frac{\theta_0}{2}(G_2-iG_1).$$

Z_0 is used to denote the specific acoustic impedance. The specific acoustic resistance and reactance R_0 and X_0 are given by

$$R_0=\rho_0 v\cos^2\frac{\theta_0}{2}hG_2, \quad X_0=-\rho_0 v\cos^2\frac{\theta_0}{2}hG_1.$$

These expressions are valid for the range of values of c^2 for which the joining factors have been obtained. Since G_1 and G_2 are functions of h and since $h=2\pi a/\lambda\sin\theta_0$, $R_0/\rho_0 v$ and $X_0/\rho_0 v$ have been plotted against $2\pi a/\lambda$ in Figs. 6

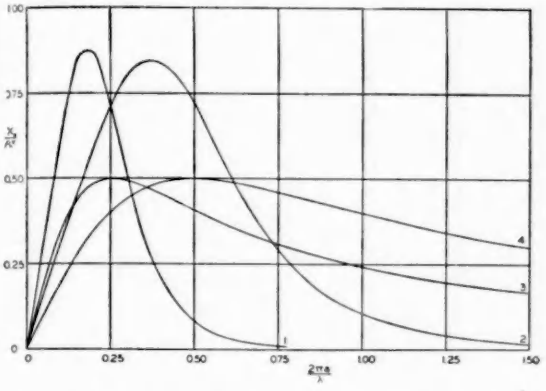

FIG. 7. Reactive components of the acoustical impedance. Curves 1, 2, 3, and 4, are for the 15° hyperbolic, 30° hyperbolic, 15° conical, and 30° conical horns, respectively.

and 7. For comparison, curves for the corresponding conical horns have been included. Corresponding hyperbolic and conical horns have the same radii at the throat and the same angular openings. For a conical horn, it may be shown by a procedure similar to that employed above that

$$\frac{R_0}{\rho_0 v}=\frac{h^2}{1+h^2} \quad \text{and} \quad \frac{X_0}{\rho_0 v}=\frac{h}{1+h^2}.$$

To investigate the behavior of R_0 and X_0 for values of h somewhat greater than those given in the figures, it is possible to make use of a second-order WKB representation of F_0.

If y satisfies the Sturm equation

$$\frac{d}{dx}(ry')+sy=0,$$

it is appropriate to assume that y may be represented by the expression $y=Pe^{\pm iQ}$ where P and Q are real functions of x. The equations which result from substituting this expression into the Sturm equation may be integrated by an iteration process. In this way, one obtains as a first approximation the ordinary WKB representation of y. The next step yields

$$Q=\int u^{\frac{1}{2}}dx, \quad P=L/r^{\frac{1}{2}}u^{\frac{1}{4}},$$

where

$$u=\frac{s}{r}-\frac{1}{4}\left[\frac{s''}{s}+\frac{r''}{r}+\frac{1}{2}\frac{s'}{s}\frac{r'}{r}-\frac{5}{4}\left(\frac{s'}{s}\right)^2-\frac{1}{4}\left(\frac{r'}{r}\right)^2\right]$$

and L is a constant of integration. These results may be applied directly to the case of F_0 by putting $r=1+x^2$ and $s=-c^2(\delta_0+x^2)$. Since F_0 represents a wave traveling outward

$$F_0=Pe^{-iQ} \quad \text{and} \quad \frac{F_0}{F_0'}=\frac{1}{P'/P-iQ'}.$$

It is readily verified that

$$\frac{P'(0)}{P(0)}=0 \quad \text{and} \quad Q'(0)=\left(-c^2\delta_0+\frac{1+\delta_0}{2\delta_0}\right)^{\frac{1}{2}}.$$

Thus

$$\frac{F_0(0)}{F_0'(0)}=\frac{i}{\left[\delta_0\left(h^2-\frac{1+\delta_0}{2\delta_0^2}\right)\right]^{\frac{1}{2}}};$$

and therefore

$$G_1=0, \quad G_2=\frac{1}{\left[\delta_0\left(h^2-\frac{1+\delta_0}{2\delta_0^2}\right)\right]^{\frac{1}{2}}}.$$

For the range of values of h for which the second-order WKB approximation to F_0 is valid

$$\frac{R_0}{\rho_0 v}=\frac{\cos^2\frac{\theta_0}{2}h}{\left[\delta_0\left(h^2-\frac{1+\delta_0}{2\delta_0^2}\right)\right]^{\frac{1}{2}}} \quad \text{and} \quad X_0=0.$$

If the variation of δ_0 be neglected over the range of h for which diffraction takes place, it follows that

$$\frac{R_0}{\rho_0 v}=\frac{h}{(h^2-1.05)^{\frac{1}{2}}} \quad \text{for the 15° horn,}$$

$$=\frac{h}{(h^2-1.23)^{\frac{1}{2}}} \quad \text{for the 30° horn.}$$

5. CONCLUSION

In case u_0, the velocity amplitude of the diaphragm, is independent of the frequency, the power radiated by a horn is proportional to the transmission coefficient $R_0/\rho_0 v$. As is evident from Fig. 6, the hyperbolic horns favor the low frequencies to a much greater extent than do

the corresponding conical ones. Since the hyperbolic horn differs in shape from the conical only in the curvature near the throat, its better performance must be attributed to that curvature. It appears that the ideal horn shape approaches that of a uniform tube near the throat. This result is supported by the fact that the exponential horn, whose uniform response characteristic is well known, has a nearly tubular throat.

The author wishes to express his gratitude to Professor J. A. Stratton who proposed this problem and contributed many helpful suggestions and to Professor S. H. Caldwell for the use of the Massachusetts Institute of Technology differential analyzer.

Graphical Determination of the Random Efficiency of Microphones

BENJAMIN BAUMZWEIGER
Shure Brothers, Chicago, Illinois
(Received January 12, 1940)

ONE of the factors indicating the directional properties of a microphone is its random efficiency,[1] i.e., the relative ability to receive sounds which are at random in direction and phase distribution. This characteristic is usually expressed as the ratio of the power translated by the directional microphone when subjected to a random sound field, to the power that would have been translated had the microphone been perfectly nondirectional. In dealing with idealized microphone structures, the R.E. may be obtained mathematically from the theoretical polar response.[2] The polar response of actual microphones is oftentimes too complex to be handled analytically, and therefore the evaluation of the R.E. is frequently made through a point-by-point summation.[3] The method described below permits direct determination of the R.E. from the polar response pattern plotted on special coordinate paper. A possible extension of this method into other fields of measurement is briefly mentioned at the end of this article.

The generalized expression for the R.E. of a microphone is a function of its polar characteristic in all directions in space.[4,5] In the usual case of a microphone which is substantially symmetrical about an axis perpendicular to the diaphragm or the plane of a ribbon, the R.E. may be expressed as a function of the polar characteristic in a plane passing through the axis of symmetry. Let this characteristic be represented by

$$f(\theta) = R_\theta / R_0 \qquad (1)$$

where R_0 is the normal incidence field response in terms of open circuit voltage per unit pressure, and R_θ is the field response for the incidence of sound from the angle θ.

Considering the microphone placed in the center of an imaginary sphere of radius a (Fig. 1) and subjected to random sounds, all directions of arrival being equally probable, the power output due to sounds arriving through a zone of width $d\theta$ at θ degrees from the axis of symmetry is proportional to $2\pi a^2 f^2(\theta) \sin \theta d\theta$. Rayleigh[6] has shown that, if the phase distribution is entirely random, the total resulting intensity in a large number of trials is the sum of the individual intensities, and therefore it follows that the total power output is proportional to

$$2\pi a^2 \int_0^\pi f^2(\theta) \sin \theta d\theta. \qquad (2)$$

In a perfectly nondirectional microphone $f(\theta) = 1$, and the solution of Eq. (2) is $4\pi a^2$; dividing this into Eq. (2) in accordance with the defini-

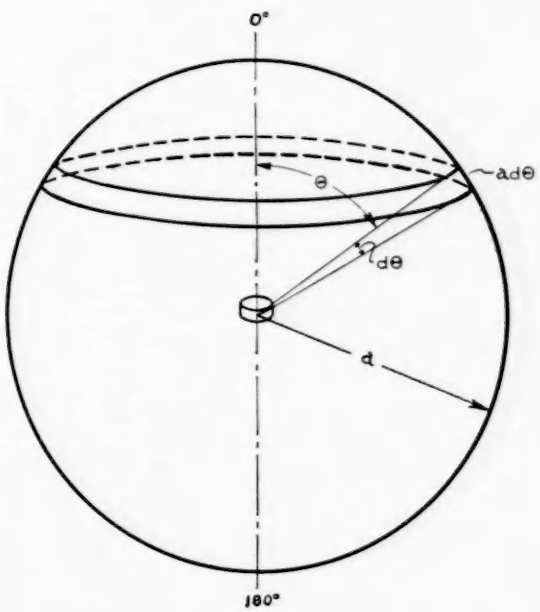

FIG. 1.

[1] Also known as the "directional efficiency."
[2] H. F. Olson and F. Massa, *Applied Acoustics* (P. Blakiston's Son & Co., 2nd edition, 1939), p. 139. See also H. F. Olson, J. Soc. Mot. Pict. Eng. **16**, No. 6 (1931), and Proc. I. R. E. **21**, 655 (1933).
[3] For example: "American Tentative Standards for Sound Level Meters," American Standards Association, No. Z24.3–1936, page 8.
[4] "American Recommended Practice for the Calibration of Microphones," American Standards Association, No. Z24.4–1938, page 7.
[5] L. J. Sivian, "Absolute Calibration of Condenser Transmitters," Bell Sys. Tech. J. **10**, 108 (1931).
[6] Rayleigh, *The Theory of Sound* (Macmillan & Co., 2nd edition, 1929), page 36.

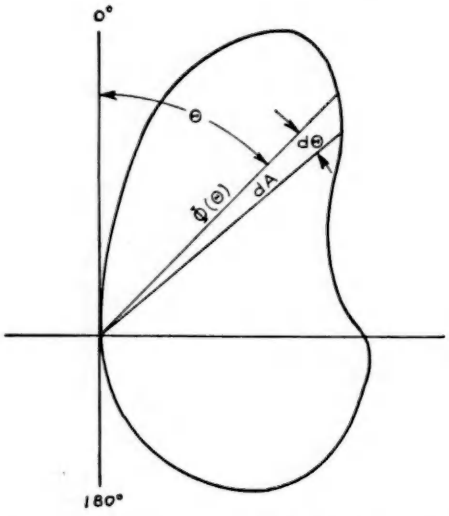

FIG. 2. Auxiliary function $\phi(\theta)$ for determination of the R. E.

tion for R.E.,

$$\text{R.E.} = \tfrac{1}{2}\int_0^{\pi} f^2(\theta) \sin\theta d\theta. \qquad (3)$$

For the purpose of graphical integration of Eq. (3), let $\phi(\theta) = f(\theta) \sin^{\frac{1}{2}}\theta$ be plotted in polar coordinates between the angles 0° and 180° as shown in Fig. 2. Then the area between $\phi(\theta)$ and $\phi(\theta+d\theta)$ is

$$dA = \tfrac{1}{2}\phi^2(\theta)d\theta = \tfrac{1}{2}f^2(\theta) \sin\theta d\theta \qquad (4)$$

and the total area enclosed by the curve becomes the solution of Eq. (3). $\phi(\theta)$ is easily plotted in terms of $f(\theta)$ through the use of a polar coordinate paper in which all the radial divisions are proportional to $\sin^{\frac{1}{2}}\theta$. An analysis sheet having this coordinate system as well as blank spaces for tabulating the results is shown in Fig. 3, together with a sample analysis for the R.E. of a typical semi-directional pressure microphone. The polar curves are obtained in the customary manner, i.e., by rotating the microphone in front of a sound source, and the percent response is plotted directly on the analysis sheet. For convenience, a length of 5 inches has been chosen to represent $f(0)$, and hence the correct R.E. is obtained by dividing 5^2 into the area enclosed by the curve, (measured in square inches by means of a planimeter). For structures which are not completely symmetrical about the 0° axis, two or more polar curves may be plotted at each fre-

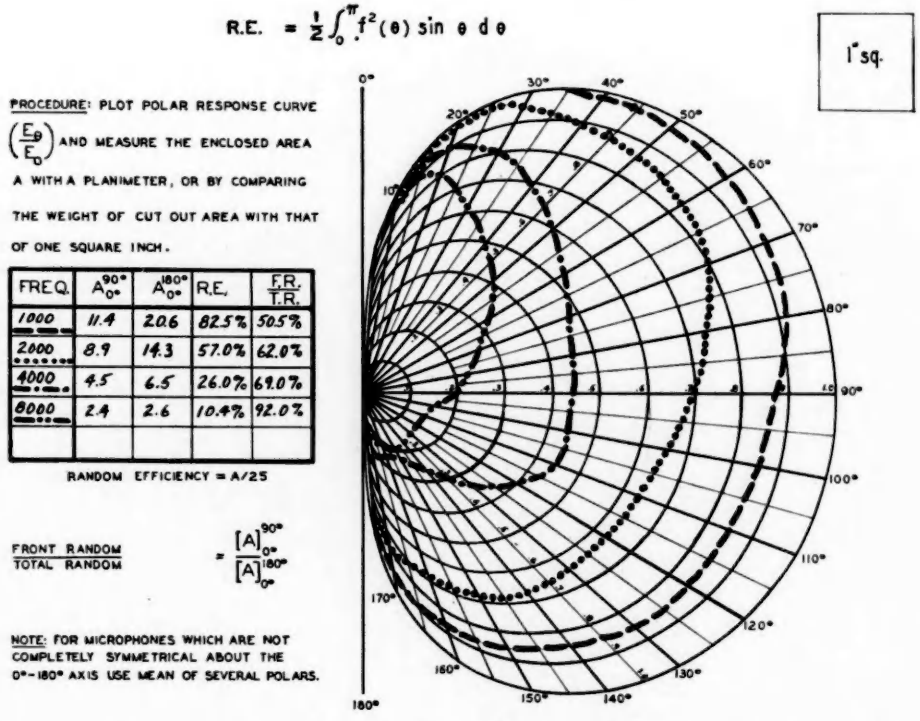
R.E. = ½ ∫₀^π f²(θ) sin θ dθ
1" sq.
PROCEDURE: PLOT POLAR RESPONSE CURVE (Eθ/E0) AND MEASURE THE ENCLOSED AREA A WITH A PLANIMETER, OR BY COMPARING THE WEIGHT OF CUT OUT AREA WITH THAT OF ONE SQUARE INCH.
FREQ. | A 90°/0° | A 180°/0° | R.E. | F.R./T.R.
1000 | 11.4 | 20.6 | 82.5% | 50.5%
2000 | 8.9 | 14.3 | 57.0% | 62.0%
4000 | 4.5 | 6.5 | 26.0% | 69.0%
8000 | 2.4 | 2.6 | 10.4% | 92.0%
RANDOM EFFICIENCY = A/25
FRONT RANDOM / TOTAL RANDOM = [A]90°0° / [A]180°0°
NOTE: FOR MICROPHONES WHICH ARE NOT COMPLETELY SYMMETRICAL ABOUT THE 0°-180° AXIS USE MEAN OF SEVERAL POLARS.
0°
30°
40°
50°
60°
70°
80°
90°
100°
110°
120°
130°
140°
150°
160°
170°
180°

FIG. 3. Sample calculation of random efficiency and front-to-total random ratio of a typical semi-directional pressure microphone.

quency through representative planes, and the results averaged.

A similar chart may be obtained in the db scale by making the radial divisions proportional to $(\log^{-1} \text{db}/20) \sin^{\frac{1}{2}} \theta$. In this manner, polar patterns obtained from instruments calibrated in db may be plotted directly on the chart for determination of the R.E.

Other directional indexes may be obtained from the polar plot in Fig. 3 in addition to the R.E. For instance, a performance index has been proposed for unilateral microphones which is expressed as the ratio of response caused by random sounds received through a hypothetical hemisphere at the front of the microphone to the total random response.[7] This index may be easily obtained from Fig. 3 by dividing the total area enclosed by the polar chart into the area enclosed between 0° and 90° azimuths.

Integrals having the form of Eq. (3) occur occasionally in connection with other applications and the described method may be adapted to their solution. Measurements of the total power radiated by a loudspeaker[8] may be mentioned as an example. Referring again to Fig. 1, if the polar radiation pattern of a loudspeaker symmetrical about the 0°–180° axis is given as p_θ dynes per square centimeter measured at the radius a cm, the power radiated through a zone of width $d\theta$ is

$$dP = \frac{2\pi a^2 p^2 \sin \theta d\theta}{\rho c} \text{ ergs per sec.,} \qquad (5)$$

where ρ is the density of air, c is the velocity of sound, cm/sec. Setting the ratio of pressure at the angle θ to that at the angle 0 equal to $f(\theta) = p_\theta / p_0$, the total power radiated becomes

$$P = \frac{4\pi p_0^2 a^2}{\rho c} \left\{ \tfrac{1}{2} \int_0^\pi f^2(\theta) \sin \theta d\theta \right\} \text{ ergs per sec.} \qquad (6)$$

The term in brackets is evaluated graphically by means of the chart in Fig. 3 in the manner described in connection with the R.E. The graphical method greatly reduces the amount of labor associated with the conventional methods of evaluating the performance of instruments having complicated polar patterns.

ACKNOWLEDGMENT

In the preparation of this article, the writer wishes to acknowledge with gratitude the advice and counsel of Ralph P. Glover, Chief Engineer of Shure Brothers.

[7] R. P. Glover, "Review of Cardioid Type Unidirectional Microphones," J. Acous. Soc. Am. **11**, 301 (1940).

[8] For example, "Standards on Electroacoustics, 1938," (Institute of Radio Engineers), p. 31.

The Attenuation of Sound in Tubes

R. ROGERS
University of Illinois, Urbana, Illinois
(Received January 17, 1940)

INTRODUCTION

IT is often necessary to calculate the attenuation of sound in a tube lined with an absorbing material. This paper presents a theory of the propagation of sound in tubes together with an approximate method for calculating attenuations. The theory here presented treats only rectangular tubes, however, the approximate expressions are equally valid for circular tubes. The results are found to agree with the previous work of Sivian[1] for low frequencies; the present analysis also covers high frequencies and a formula is obtained.

The first part of this paper presents the theory of the propagation of sound in tubes, and the second part presents experimental confirmation for the methods of calculation proposed.

THEORY

In this work the wave equation is solved subject to suitable boundary conditions on the walls of the tube. This result gives an expression for the attenuation of sound passing down the tube. If the tube is rectangular of cross-sectional dimensions $2b$ and $2d$, and if the origin is chosen in the middle of the tube, the conditions to be satisfied on the walls are:

$$\begin{aligned} p/\dot{\eta}&=Z_y \quad \text{at} \quad y=b,\\ p/\dot{\eta}&=-Z_y \quad \text{at} \quad y=-b,\\ p/\dot{\xi}&=Z_z \quad \text{at} \quad z=d,\\ p/\dot{\xi}&=-Z_z \quad \text{at} \quad z=-d, \end{aligned} \tag{1}$$

where p is the excess pressure, $\dot{\eta}$ and $\dot{\xi}$ are the particle velocities in the y and z directions, and Z_y and Z_z are the acoustic impedances of the walls perpendicular to the y and z axes.

If the driving motion is sinusoidal of frequency $\omega/2\pi$, a suitable solution of the wave equation gives for the velocity potential

$$\varphi=\Sigma\Sigma A_{rs}(x)\cosh m_r y\cosh n_s z e^{-i\omega t},$$

where m_r and n_s are constants to be determined, and where $A_{rs}(x)$ is an undetermined function. Putting this expression for φ in the wave equation gives

$$A_{rs}=B_{rs}e^{i\rho_{rs}x}+C_{rs}e^{-i\rho_{rs}x}, \tag{2}$$

where B and C are constants depending on the boundary conditions at the ends of the tube, and where

$$\rho_{rs}^2=\frac{\omega^2}{c^2}+m_r^2+n_s^2. \tag{3}$$

The quantity of interest is the rate of decay with distance along the tube. This is represented in Eq. (2) by the imaginary part of ρ_{rs} for each mode of vibration. This may be determined from Eq. (3) by knowing m_r and n_s.

The problems of determining m_r and n_s are exactly equivalent so that it is only necessary to treat the one. Thus using the first boundary condition m_r is determined by

$$m_r\tanh m_r b=-\frac{i\omega\rho_0}{Z_y}, \tag{4}$$

where ρ_0 is the density of air. This equation has an infinity of solutions. These may be obtained by letting $x+r\pi i=m_r b$ and solving the equation

$$(x+r\pi i)\tanh x=-\frac{i\omega\rho_0 b}{Z_y}=M, \tag{5}$$

where r is any integer. The solution may be carried out by expanding $\tanh x$ in series and inverting the series by Burman's theorem.[2] This gives two solutions depending on whether r is 0 or some other integer. When $r=0$

$$x^2=M+\frac{1}{3}M^2+\frac{4}{45}M^3+\cdots, \tag{6}$$

and when $r>0$

$$x=\frac{M}{r\pi i}-\frac{1}{r\pi i}\left(\frac{M}{r\pi i}\right)^2-\left(\frac{2}{3}\frac{1}{r\pi i}-\frac{1}{3}\right)\left(\frac{M}{r\pi i}\right)^3+\cdots. \tag{7}$$

[1] L. J. Sivian, "Sound Propagation in Ducts," J. Acous. Soc. Am. 9, 135 (1937).

[2] E. T. Whittaker and G. N. Watson, *Modern Analysis* (Cambridge, 1927).

If the lining is a porous material the acoustic impedance may be assumed real and equal to R.[3] Further if any practical use is to be made of the equations they must be simplified. To do this it is assumed that $|M| \ll 1$ and that $|M/r\pi i| \ll 1$. Thus if $m_r = m_r' + m_r'' i$ Eq. (6) is

$$m_0' m_0'' = -\omega\rho_0/2bR_y, \tag{8}$$

where only the first term of the series is retained. In the same way Eq. (7) gives

$$m_r' m_r'' = -\omega\rho_0/bR_y, \tag{9}$$

where $r = 1, 2, 3\cdots$.

If the notation is used that a single prime refers to the real part of that quantity to which it is attached and a double prime refers to the imaginary part, ρ_{rs} may be obtained from Eq. (3) as

$$\rho_{rs} = -\frac{\omega}{c}\left[1 + \frac{1}{2}\frac{c^2}{\omega^2}(m_r^2 + n_s^2) + \cdots\right]$$

by expanding by the binomial theorem for the case where

$$\omega^2/c^2 > |m_r^2 + n_s^2|$$

and ρ_{rs}'' is

$$\rho_{rs}'' = -\frac{1}{2}\frac{c}{\omega}(2m_r' m_r'' + 2n_s' n_s'') + \cdots. \tag{10}$$

Knowing $m_r' m_r''$ and $n_s' n_s''$ from Eqs. (8) and (9) the value of ρ_{rs}'' may be easily calculated. If terms of a higher order are used and if at the same time the simple approximations (8) and (9) are used, terms in M^2 will be introduced. Since these have already been neglected in obtaining Eqs. (8) and (9) this further refinement is not valid. If the value of $m_0' m_0''$ be used in the series (10) the same expression for ρ_{rs}'' is obtained as that given by Sivian.[1] His result applies to the case in which only the lowest mode is present.

A condition has been stated under which the expansion given in Eq. (10) is valid. It is of interest to inquire what happens when the opposite is true that is when $\omega^2/c^2 < |m_r^2 + n_s^2|$. Physically this means that the sound strikes the wall many times while traveling down the tube, whereas the previous condition required that the direction of propagation make only a small angle with the axis of the tube. Thus it is intuitively reasonable that the second class of waves would be damped out much more rapidly than the first class. This may be rigorously justified except when $\omega^2/c^2 < |m_0^2 + n_0^2|$. This last condition includes only a small number of cases since for it to be true ω must be very small or the impedance of the walls must be nearly equal to that of air.

Equation (10) in conjunction with Eqs. (8) and (9) then represents the decay of sound in the tube save in a very short range near the source.

Since the number of modes of vibration present in the tube depends upon the source, this factor must be considered in calculating the attenuation coefficient. Thus in an infinite tube the velocity potential is given as a function of x alone as

$$A = \Sigma A_{rs} = A_{00}e^{-\rho_{00}'' x} + A_{01}e^{-\rho_{01}'' x} + A_{10}e^{-\rho_{10}'' x} + A_{11}e^{-\rho_{11}'' x}, \tag{11}$$

where A_{11} represents the part of the total velocity potential contributed by all modes of vibration for which r and s are both greater than 0. The values of the various attenuation constants are calculated by substitution in Eq. (10) from Eqs. (8) and (9). These are summarized below.

$$\begin{aligned}
\rho_{00}'' &= \rho_0 c\left(\frac{1}{2R_y b} + \frac{1}{2R_z d}\right),\\
\rho_{01}'' &= \rho_0 c\left(\frac{1}{2R_y b} + \frac{1}{R_z d}\right),\\
\rho_{10}'' &= \rho_0 c\left(\frac{1}{R_y b} + \frac{1}{2R_z d}\right),\\
\rho_{11}'' &= \rho_0 c\left(\frac{1}{R_y b} + \frac{1}{R_z d}\right).
\end{aligned} \tag{12}$$

For a rigorous solution it is only necessary to expand the source function in a series of nearly orthogonal functions $\cosh m_r y \cosh n_s z$ to find

[3] For a treatment of the case in which the acoustic impedance cannot be considered real, see the recent paper by P. M. Morse, J. Acous. Soc. Am. **11**, 205 (1939). In the experimental work performed by the author it was found to be satisfactory to assume the impedance real when the lining was a porous material. In this connection note the paper by F. V. Hunt, L. L. Beranek and D. Y. Maa, J. Acous. Soc. Am. **11**, 80 (1939), and reference 5.

the coefficients A_{rs}. Since the source function is seldom known exactly it is not of practical interest to pursue this line of attack further. For the experimental work performed by the author the assumption was made that at low frequencies the attenuation was given by ρ_{00}'' and that at high frequencies it was given by ρ_{11}''. The reasons for this assumption will appear in the next paragraph.

Since only those modes for which $|m_r^2+n_s^2| < \omega^2/c^2$ are of importance in transferring energy along the tube, it is possible to eliminate at once from consideration those which do not satisfy this condition. The approximate value of m_r^2 and n_s^2 may be obtained at once from Eq. (7). The major term in this equation is $r\pi i$. Thus as a first approximation

$$|m_r^2+n_s^2| \backsimeq \left(\frac{r\pi}{b}\right)^2+\left(\frac{s\pi}{d}\right)^2. \tag{13}$$

In the solution for φ only the hyperbolic cosine has been used. A solution containing the hyperbolic sine is, however, equally valid. In this case the approximate solution is the same except that r and s are odd multiples of one-half. The complete solution is therefore that already obtained with r and s given as any integral multiple of one-half. Thus the only important modes are those for which

$$\left(\frac{r\pi}{b}\right)^2+\left(\frac{s\pi}{d}\right)^2 < \frac{\omega^2}{c^2}, \tag{14}$$

where

$$r=\tfrac{1}{2}, 1, \tfrac{3}{2}\cdots,$$
$$s=\tfrac{1}{2}, 1, \tfrac{3}{2}\cdots.$$

When

$$\frac{\omega^2}{c^2} > \left(\frac{\pi}{b}\right)^2+\left(\frac{\pi}{d}\right)^2$$

the value of ρ_{rs}'' is definitely established as ρ_{00}''. When

$$\frac{\omega^2}{c^2} > \left(\frac{\pi}{b}\right)^2+\left(\frac{\pi}{d}\right)^2$$

any or all of the four modes may be set up. If the source is a small loudspeaker, the sound tends to spread out in all directions forming a curved wave front. In this case it is natural to expect that the greater part of the energy would lie in those modes having a direction of propagation which makes a small angle with the axis of the tube. These modes all have a constant decay factor, ρ_{11}''. For frequencies between these two limits there is no simple criterion for selecting the modes. The preceding argument is not a rigorous proof for the criteria used. The best proof is to be found in the experimental results of the next section.

If the diameter of the sound source becomes comparable with the wave-length of the sound radiated, the radiation is confined to a beam and does not strike the walls of the tube. In this case the results above do not hold as both Sivian[1] and the author have observed.

In practice it is more convenient to express the decay in intensity in decibels per foot. Thus for an infinite tube the intensity at any distance x from the source is

$$I=I_0 10^{-\alpha x/10}, \tag{15}$$

where α will hence forth be called the attenuation coefficient. If D and D_0 represent the intensities I and I_0 in terms of decibels

$$\alpha x = -10 \log_{10} (I/I_0) = D_0 - D. \tag{16}$$

Since the intensities are proportional to the squares of the pressures or of the velocity potentials, I is also given by

$$I=I_0 e^{-2\rho_{rs}'' x}$$

for that particular mode represented by r and s. Comparing this with Eq. (15)

$$\alpha_{rs}=20\rho_{rs}'' \log_{10} e = 8.7\rho_{rs}''.$$

For high frequencies all values of ρ_{rs}'' are equal to ρ_{11}''. Thus

$$\alpha_{11}=8.7\rho_0 c(1/R_y b+1/R_z d). \tag{17}$$

To obtain the other values for α it is only necessary to replace the quantity in parentheses by the corresponding quantities given in Eqs. (12).

When all walls are covered with the same material $R_y=R_z=R$ and Eq. (17) becomes

$$\alpha_{11}=\frac{8.7\rho_0 c}{R}\frac{b+d}{bd}=\frac{8.7\rho_0 c}{R}\frac{P}{A}, \tag{18}$$

where P is the perimeter and A is the cross section. From this it is evident that for a given

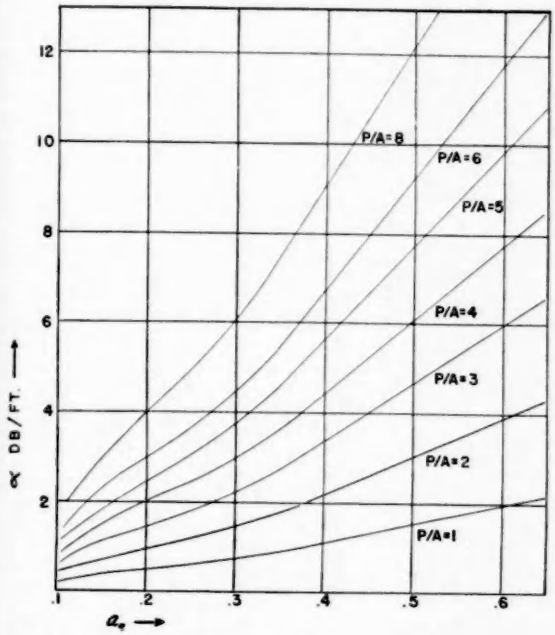

FIG. 1. The attenuation coefficient α as a function of the normal absorption coefficient a_0 for several values of the ratio of the perimeter to the area. These values are valid only when all walls of the tube are covered with the same material and when the frequency is high.

lining the attenuation is directly proportional to the ratio of P to A.

The attenuation coefficient may also be expressed as a function of the normal absorption coefficient, a_0, instead of the acoustic impedance. Morse[4] gives a relation between a_0 and R which reduces to

$$\frac{R}{\rho_0 c}=\frac{1+(1-a_0)^{\frac{1}{2}}}{1-(1-a_0)^{\frac{1}{2}}}. \tag{19}$$

Substituting this value for R in Eq. (18) gives an equation valid at high frequencies. This is

$$\alpha_{11}=8.7\frac{P}{A}\frac{1-(1-a_0)^{\frac{1}{2}}}{1+(1-a_0)^{\frac{1}{2}}}. \tag{20}$$

For convenience of reference α is given in Fig. 1 as a function of a_0 for several values of P/A.

EXPERIMENTAL

The theory presented in the previous section lends itself to experimental confirmation in two ways. First, Eq. (18) shows the attenuation coefficient to be proportional to P/A. Second, Eq. (20) gives a method of calculating the normal absorption coefficient from measurements of the attenuation coefficient.

To measure attenuation coefficients, it is convenient to make use of a tube sufficiently long that the reflected wave is negligible. In the tests made to check the theory, a tube 32 feet long and 1 foot square was used. One dimension was made variable in order to make measurements for several values of P/A. To insure that the reflected wave is negligible it is only necessary that the intensities in decibels when plotted as a function of x lie on a straight line. This has been done in Fig. 2. Here it is seen that the necessary straight line is attained over a considerable length of the tube. Caution must be observed not to make measurements close to the source or to the open end.

In making measurements of this type it is necessary that the sound field be undistorted by the measuring device. To insure this a very small microphone was used. The arrangement for measuring the intensities is shown schematically in Fig. 3.

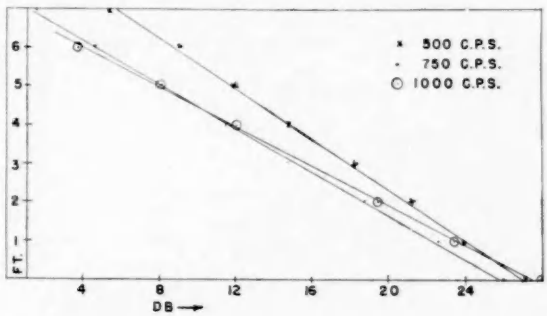

FIG. 2. Experimental values of the intensity showing the linearity of decay with distance.

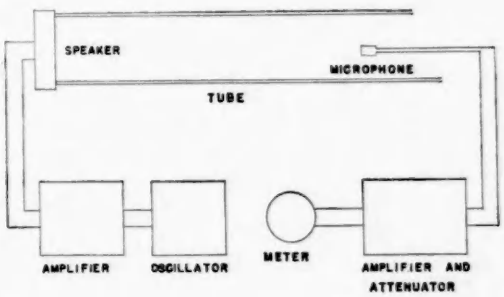

FIG. 3. Sound intensity measuring apparatus.

[4] P. M. Morse, *Vibration and Sound* (McGraw-Hill, 1936).

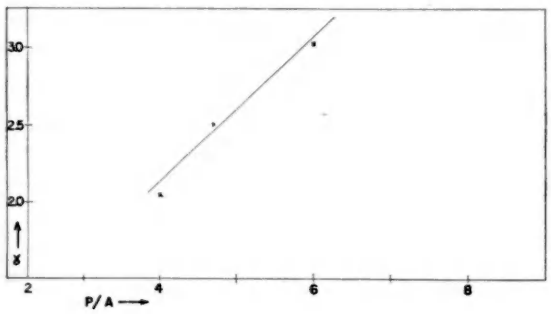

FIG. 4. Measured values of α showing the proportionality of α and P/A.

Figure 4 gives a plot of α as a function of P/A. The straight line that results shows the proportionality between the absorption coefficient and P/A. The results of the other method of checking the theory are shown in Fig. 5. Here the horizontal coordinate represents the magnitude of the absorption coefficient of a material at a certain frequency when measured by the standing wave method or by the direct reflection method of Willig.[5] The vertical coordinate is a measure of the magnitude of the absorption coefficient calculated from attenuation coefficients by the use of Eq. (20). Since both methods should give the same coefficient the points should fall along a straight line making an angle of 45° with the axes. The departures of the points from this line as shown in Fig. 5 are within experimental error.

These experiments were carried out at frequencies to which Eq. (20) is applicable. Confirmation of the low frequency formula contained in the first of Eqs. (12) has already been given by Sivian.[1]

[5] F. J. Willig, "Comparison of Sound Absorption Coefficients by Different Methods," J. Acous. Soc. Am. **10**, 293 (1939).

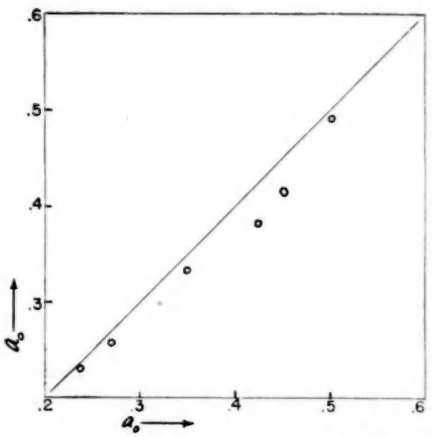

FIG. 5. Vertical values of a_0 measured by the standing wave method. Horizontal values calculated from measurements of attenuation coefficients.

CONCLUSION

In this paper a method of calculating attenuation coefficients has been given which is summarized in Eqs. (12). While these are only first approximations, experimental evidence has been presented to show their validity for a large number of cases met with in practice. A number of cases of theoretical interest and some which may be of practical importance have not been treated here. The value of introducing the further complications necessary for the treatment of these cases must be left for the decision of the practical engineer. Also, further experimentation is necessary to determine under what conditions it is possible to assume the impedance of the walls to be real.

In conclusion the author wishes to express his appreciation of the interest and assistance of Professors F. R. Watson and H. M. Mott-Smith, and his indebtedness to Dr. F. J. Willig for most of the measurements of absorption coefficients used here.

Errata: On the Theory of Fluctuations in the Decay of Sound[1]

R. Clark Jones
Research Laboratory of Physics, Harvard University, Cambridge, Massachusetts

BECAUSE of an error in copying, three of the results quoted in the brief section, The Transition (p. 331), are incorrect. The three expressions $(\pi p_0 k N_0)^2$, $\pi p_0^2 k N_0$, and $1/(\pi k N_0)$ should be $(\frac{1}{2} p_0 k N_0)^2$, $\frac{1}{2} p_0^2 k N_0$, and $2/(k N_0)$, respectively.

In stating the fundamental expression, Eq. (1), upon which the subsequent development was based, the author referred to two places where a derivation of the equation could be found, but he omitted to mention that the relation was first obtained by Kluyver.[2]

[1] R. Clark Jones, J. Acous. Soc. Am. **11**, 324 (1940).
[2] J. C. Kluyver, Proc. Section of Sciences, K. Akad. van Wet. te Amsterdam **8**, 341 (1905–06).

Letter to the Editor

"Bells"

THE Editor has received from Jan Arts[1] a note dealing with the tone which he has found[1] in a number of small bells, and which he has called an "acoustic tone." Arts now states that other persons interested in bells have also found this tone, and he suggests that it may result from the simultaneous sounding of the second and third partials of the bell. In the cases which Arts has examined he finds that the relative frequencies of the acoustic tone, the second partial, and the third partial are approximately either 4 : 5 : 6 or 3 : 4 : 5, and he states that on a piano or harmonium, and also with tuning forks, he has been able to hear a tone which has a relative frequency not far from 4 when 5 and 6 are sounding together, and a tone with relative frequency 3 when 4 and 5 are sounding together.

[1] Jan Arts, J. Acous. Soc. Am. **9**, 344 (1938).

Acoustical Society News

THE Spring meeting of the Acoustical Society will be held at the National Bureau of Standards in Washington, D. C. on April 29 and 30. There are to be several invited papers on the program relating to sound level measurements, particularly measurements at very high intensities.

After the dinner on Monday evening a trip is planned through the newly constructed Naval Model Basin Testing Laboratory, located at Carderock, Maryland, just outside of Washington. This building will be of special interest to members of the Society because of its remarkable acoustic properties.

The Wardman Park Hotel located near the National Bureau of Standards has been selected for hotel headquarters.

Translation Service

Members who run across important German or French acoustical papers which they would like to have translated and reviewed can arrange to have such reviews published in the section of our Journal entitled "Reviews of Contemporary Papers" by writing to Dr. Leo L. Beranek, Cruft Laboratory, Harvard University, Cambridge, Massachusetts. Dr. Beranek has been preparing that section of the Journal most capably for a number of months and now offers this additional service to members without charge.

New Members—Acoustical Society of America

Ivan C. Clement,
Submarine Signal Company,
160 State Street,
Boston, Massachusetts.

J. W. Clise, (Reinstated)
Asbestos Supply Company,
First Avenue at Jackson,
Seattle, Washington.

Francis Xavier Connelly,
136 Spring Street,
Rochester, New York.

Frank M. Davis,
Collins Radio Company,
2920 First Avenue,
Cedar Rapids, Iowa.

Cyril Manton Harris,
Department of Physics,
University of California,
Los Angeles, California.

Chesney R. Moe,
Department of Physics,
San Diego State College,
San Diego, California.

Walter W. Mueller,
35–21 32nd Street,
Long Island City, New York.

Rudolph Henry Nichols, Jr.,
Owens Illinois Glass Company,
Alton, Illinois.

John Sangster,
Celotex Ltd., North Circular Road,
London, N.W. 10, England.

Robert Sonnenschein,
4518 Woodlawn Avenue,
Chicago, Illinois.

Robert Lynn Thompson,
208 17th Street, N.E.,
Cedar Rapids, Iowa.

Francis Michael Wiener,
14 Remington Street,
Cambridge, Massachusetts.

Deceased

E. E. Free, Fellow
Jack Ernest Constable, Member

Fellows	132
Members	593

Total Membership, 725

Current Publications on Acoustics

F. A. FIRESTONE

147 East Physics Building, University of Michigan, Ann Arbor, Michigan

Reviews of Contemporary Papers

MEMBERS who have occasion to translate important papers from foreign journals will be doing a real service to those members who do not read the foreign languages easily, if they will prepare a review of such articles for publication in this section.

These reviews should be of the nature of a lengthy abstract (500 to 1500 words) rather than a critique or appraisal, and should attempt to set forth in this limited space as much of the original author's contribution as possible. One or two figures may be included if desired.

Particular articles in French or German periodicals will be translated and reviewed in these columns by request. Address requests to Leo L. Beranek, Cruft Laboratory, Harvard University, Cambridge, Massachusetts.

On the Perception of Vibrations. GEORGE V. BÉKÉSY, Akustische Zeits. **4**, 316 (1939).—Audiofrequency vibrations can be perceived in other portions of the body than the familiar auditory mechanism. This paper presents measurements made on the transmission of vibrations through the body and over the surface of the skin, and discusses the physiological mechanism for the perception of them.

In the first experiment described, a vibrating probe with a surface area of one cm² was held against a point of the inside lower arm near the elbow, and a very light electrodynamic microphone used to detect the vibrations perpendicular to the skin at points between the source and the wrist. For a frequency of 800 cycles the vibrations were damped 0.7 neper/cm. for relaxed muscles (one neper =8.7 db), and 0.5 neper/cm for taut muscles. At 200 cycles the damping decreased to 0.3 neper/cm. Also, near the wrist joint a very noticeable increase in amplitude of vibration occurred. Békésy explains this increase by simple mechanical considerations and correlates the effect with the subjective observation that strong vibrations are usually felt most at the joints of the arms and legs.

The author next investigates the propagation of low frequency vibrations through the human body. A small electric motor was equipped with an eccentric mass at each end of its rotor shaft such that it executed an amplitude of vibration of 1.5 mm. The oscillations at the surface of the body were picked up by a very light rubber diaphragm and transmitted through a rubber hose to a diaphragm-mirror type manometer. With the motor held in the hand, pressure amplitude measurements were made along the arm, neck and head. The amplitudes were damped by a total of 1.7 nepers along the arm, with an added abrupt decrease of 1.6 nepers occurring at the shoulder. The total reduction between the hand and the top of the head was 4.7 nepers. Measurements were then made with the subject standing on a small platform affixed to the motor. In this case the amplitude of vibration decreased almost logarithmically with distance along the body; the total reduction amounting to 3.4 nepers between the feet and the head. Transmission of audiovibrations through the vertebrae was next checked using six vertebrae loaded at the top with a mass of two kg. Almost no damping was found for frequencies below 500 cycles. This unexpected result in the audiorange explains, perhaps, why large vibrations frequently damage the auditory organs.

It is a controversial question in physiology as to whether the perception of audiovibrations takes place in the skin, bones, or muscles. Meissner's[1] famous experiment in which the finger was immersed in mercury under static pressure showed that for the sense of touch only a pressure gradient could produce a sensation. Fig. 1 shows Békésy's experimental arrangement for determining whether this is also true for alternating pressures. At 200 cycles the diaphragm was vibrated with a pressure of 70 g, r.m.s., weight per cm². No vibration was felt anywhere along the finger except in the vicinity of the rubber ring and also in the hand and at the elbow. This experiment proves that, similar to Meissner's result, a pressure gradient is necessary for a sensation of vibration.

By static deformations the author showed that the skin and muscles together could not be considered as an isotropic medium, but that the skin had larger stiffness than the muscles. Because of this difference, relative dis-

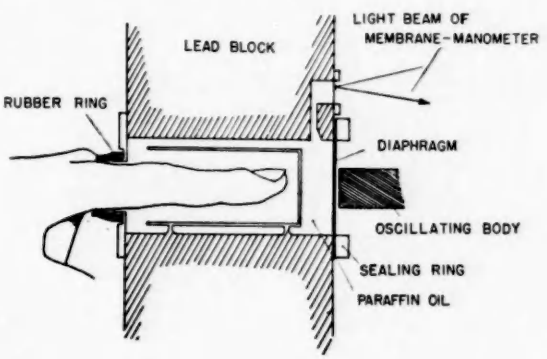

FIG. 1. Meissner experiment for vibration perception.

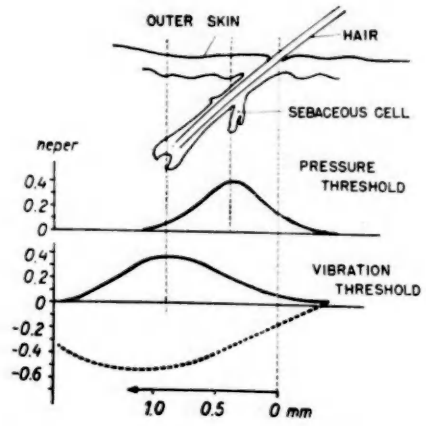

FIG. 2. Space separation of pressure and vibration perception along the outer skin.

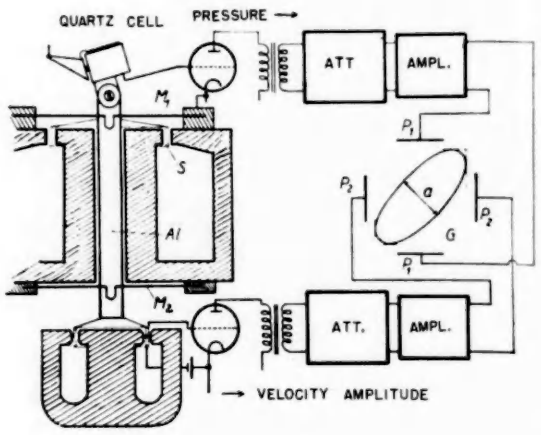

FIG. 3. Apparatus for measuring mechanical impedance.

placements take place between the skin and the muscles and these seem to play a large part in the perception of vibrations. This was further demonstrated by cementing the end of a vibrating probe to the skin of the ball of the thumb. The sensation of vibration increased both when the skin was deformed inward and when deformed outward as compared to the condition of zero static pressure. Larger relative motions of the skin and muscle occur in the two extreme positions. If the skin is not fastened to the muscle underneath, such as on the inner arm, the perception of vibration decreases greatly when the skin is bowed outwards.

For cases in which no nerves of the skin were present, due to an operation, the muscles beneath seemed to perceive vibrations, but at a lower pitch than for normal skin conditions. Even in the case of normal skin the two kinds of perception were segregated. If a two-cm² probe vibrating at about 200 cycles was pressed against a fairly large muscle, the initial skin perception would transfer to a muscle perception of lower pitch because of skin fatigue after two to four seconds.

It is known that points sensitive to static pressure exist in the vicinity of hair roots. Frey,[2] using hairs fastened to a vibrating probe, convinced himself that points of maximum pressure sensitivity and maximum vibration sensitivity coincided and were near the hair roots. He concluded that the same nervous system was effective in both cases. Békésy found, measuring on the skin of the outer arm, that points of maximum pressure sensitivity and points of maximum vibration sensitivity did not coincide. His experimental evidence pointed to the fact that one set of nerves in the sebaceous (oil) gland was sensitive to static pressures, while another set near the root (papilla) of the hair was sensitive to vibrations. The presence of these nerves has been shown anatomically. Furthermore, a hair could be so extracted that only pressure perception was left. Fig. 2, drawn in the plane of the hair, shows threshold curves for pressure and vibration perception in the vicinity of a hair. The dotted curve was obtained for the case of an extracted hair.

Knudsen[3] and others have investigated the frequency characteristic of the vibrational threshold for the finger. Knudsen found that a significant maximum in sensitivity was obtained for frequencies near 200 cycles, but did not answer the question as to whether this was a resonant effect or whether it occurred because the finger, on account of its relatively small mass, followed the vibrating body at lower frequencies with a corresponding decrease in sensitivity. Before answering this, the author found it necessary to determine as to whether the perception of vibration at the finger is due to the alternating pressure or the displacement. One way of making this determination was to measure the vibration threshold of two portions of skin which have different mechanical impedances, knowing the displacement amplitude. If in addition the impedance is known then the pressure amplitude can be found and the dependence of vibration threshold on pressure or displacement can be established. For example, if the vibration thresholds, in terms of displacement, for two such portions of skin are obtained as a function of frequency, and are of the same shape, then the threshold is displacement controlled.

To measure mechanical impedance the arrangement shown in Fig. 3 was devised. The diaphragms M_1 and M_2

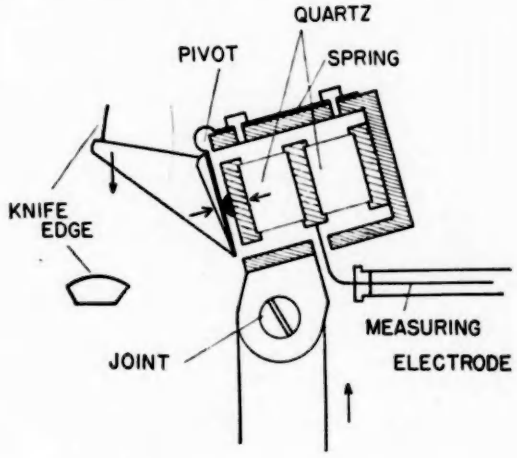

FIG. 4. Pressure measuring cell.

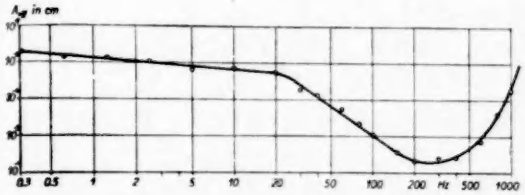

FIG. 5. Threshold curve for lateral displacement of the skin of the finger tips.

supported an aluminum rod *Al* which in turn was driven by the coil *S*. The amplitude of vibration was controlled by the current through *S*, which was so adjusted as to keep the voltage output of the electromechanical transducer, shown below, a constant. This means that the velocity was held constant, The knife edge of the quartz pressure cell[4] (Fig. 4) was pressed against the skin; the voltage output of the cell being led through an attenuator and amplifier to one pair of plates of a cathode-ray tube. The output of the transducer was led through an attenuator and amplifier to the other two plates. Since the velocity amplitude was held constant, the horizontal deflection was always a constant *G*, say. The vertical deflection was then a measure of the magnitude of the impedance. The phase angle could be determined by adjusting the attenuator connected to the vertical plates until the vertical and horizontal deflections were equal. The resulting ellipse was always in the same quadrant and the phase was given by the formula,

$$\sin \phi/2 = a/\sqrt{2}G \qquad (1)$$

or to within three percent,

$$\phi = 0.97(a/G)\cdot 90°. \qquad (2)$$

By inclining the pressure head, the voltage generated by the mass of the electrodes was balanced against that generated by the mass of the lever arm. The unit was calibrated by adding a known mass to the lever arm.

Measurements made with the finger nail pressed against a mass, showed that the front of the finger tip was a spring with a power factor of not over 0.26. The ball of the thumb, the upper arm muscle, and the free finger tip were nearly pure frictional resistances between 50 and 250 cycles. It was found that if a one-half-cm² flat probe were used in place of the knife edge the vibration threshold of sensation was determined by the velocity (or displacement). For the knife edge, where the pressure was large, both pressure amplitude and displacement amplitude seemed important.

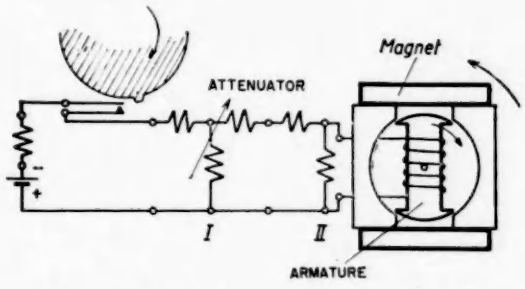

FIG. 6. Apparatus for measuring subjective strength-of-vibrations.

In the case of Knudsen's experiment, finger tip resonance was definitely not found, the decrease in sensitivity at low frequencies being attributed to the low mass of the finger. Fig. 5 is a threshold curve obtained with the aluminum rod held between the finger and the thumb. The measurements led to the same results for the upper arm and thigh, except that the threshold curve was shifted an octave lower in frequency, and the threshold was four times increased.

A subjective strength-of-vibration meter was developed as shown in Fig. 6. Since most vibrational disturbances enter the body through the feet, the instrument furnishing the standard signal was designed to be held in the hand; the hand being the most remotely coupled portion of the body. A pulse of current from the battery caused the armature to rotate violently in the magnetic field. This motion was accompanied by a motion in the opposite direction of the magnet and housing, and was transmitted through a small rod to the hand. A spring returned the armature to its rest position after each pulse. The intensity of these torsional vibrations was controlled by the attenuator. The frequency of pulses used was 10 per second and the current impulse duration 0.025 second. This type of vibration has a character easy to distinguish from most vibrational disturbances such as that of automobiles. Preliminary measurements indicated that vibrational loudness is not a measure of how a certain vibration will interfere with other operations of the body; for example, hand writing.—L. BERANEK.

[1] Meissner, Z. Rat. Med. (7), **3**, 99 (1859).
[2] M. v. Frey in J. Jadassohn, *Handbuch der Hautkrankheiten*, Vol. I/2.
[3] V. O. Knudsen, J. Gen. Psychol. **1**, 320 (1928).
[4] G. v. Békésy, Akustische Zeits. **4**, 113 (1939); L. Beranek, J. Acous. Soc. Am. **11**, 259 (1939).

On Sound Propagation at Grazing Incidence to Porous Materials. K. SCHUSTER, Akustische Zeits. **4**, 335 (1939).—Using the same treatment as that for the propagation of electric waves along a conducting plane, and (for a tube) along conducting wires, the author treats theoretically the propagation of sound waves over porous materials.

The treatment is restricted to infinitely thick porous sheets. For the case of a plane surface, a simple point source of sound energy is assumed in the surface. Following Sommerfeld,[1] the total wave field is composed of three parts:

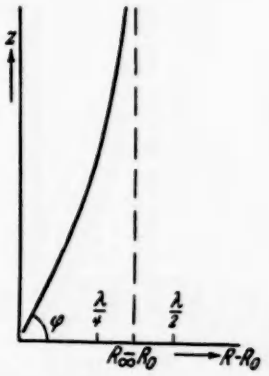

FIG. 1. Shape of the wave front above the porous material.

1. The space wave I: It is propagated out into the air with a wave number k_1 and produces in the material a disturbance known as a "head wave," because of the slower sound velocity in the material.

2. The space wave II: It is propagated out into the material with a wave number k_2 and produces a disturbance in the air such as that produced by total reflection.

3. The surface wave: This is propagated along the surface of separation with a still different wave number and is exponentially damped in both directions perpendicular to the bounding surface.

Far removed from the source, in medium I, only the space wave I is important. Let the coordinates of a point in medium I be R and z, where R is the radial distance from the source and z is the perpendicular distance from the bounding surface. Then if $R \gg z$ and the frequency is not too low, Schuster derives the velocity potential in the air as being:

$$\psi_1 \propto \left[\frac{\frac{\rho_0}{\rho}\left(1-j\frac{r}{\omega\rho}\right)}{\sigma\left(j\frac{\rho_0 r}{\omega\rho^2}-\left(\frac{\rho_0}{\rho}-1\right)\right)^{\frac{1}{2}}}+k_1 z \right] \frac{e^{-jk_1R}}{k_1^2R^2}, \qquad (1)$$

where r is the specific resistance to flow, σ is the porosity, ρ is the density of air, ρ_0 is the apparent increased density of air in the pores of the material, c is the velocity of sound, and ck_1 is the angular frequency.

Equation (1) shows that the sound pressure amplitude decreases as $1/(R)^2$ and not as $1/R$ as for the ordinary spherical case. Also the sound pressure decreases as the material is approached ($z \to 0$). These results have been checked experimentally.[2,3]

Under the condition that $\rho_0 r/\rho^2\omega \gg (\rho_0/\rho - 1)$, the equation of an equal phase surface for R very large is given as:

$$z = \frac{\rho_0}{\rho\sigma\left(2\frac{\rho_0 r}{\rho^2\omega}\right)^{\frac{1}{2}}k_1} \times\left\{\left(\frac{r}{\omega\rho}+1\right)\operatorname{cotg}\,(K-k_1R)+\left(\frac{r}{\omega\rho}-1\right)\right\}, \qquad (2)$$

where K is some constant. Fig. 1 is obtained from (2) for the special case of $\rho_0/\rho = 1.5$, $\sigma = 1$, $r/\omega\rho = 2$. ϕ in this example is 64°. This checks closely an experimental result of $\phi = 67°$ by Janovsky and Spandöck at a frequency of 200 cycles.

At larger distances from the bounding surface in medium I the author derives an infinite series form for the velocity potential which gives the result that the sound pressure decreases as the absorbing surface is approached and the rate of decrease is greater as the surface is made more highly absorbing.

The surface wave along the plane porous sheet is also discussed. In this case the author derives the following equation for the wave number:

$$\gamma = k_1\left(\frac{\sigma^2k_1^2k_2^2-k_2^4}{\sigma^2k_1^4-k_2^4}\right)^{\frac{1}{2}}. \qquad (3)$$

For the special case of $\rho = \rho_0$, $\sigma = 1$, $r/\rho\omega \gg 1$, (3) becomes,

$$\gamma = k_1\left(1-j\frac{\rho\omega}{r}\right)^{\frac{1}{2}}, \qquad (4)$$

which gives as a damping constant along the wall the value,

$$\beta = k_1\cdot(\rho\omega/2r). \qquad (5)$$

For the case of surface waves along a cylindrical tube with porous walls, Schuster derives the result that the damping exponent along the tube's axis is given by:

$$\beta = \frac{\sigma}{b\left(2\frac{\rho_0 r}{\rho^2\omega}\right)^{\frac{1}{2}}} = \frac{a}{4b}, \qquad (6)$$

where b is the radius of the tube and a is the normal incidence absorption coefficient. No experimental confirmation of this is offered.—L. BERANEK.

[1] A. Sommerfeld, Ann. d. Physik **4** (28), 665 (1909).
[2] G. v. Békésy, Zeits. f. tech. Physik **14**, 6 (1933).
[3] Janovsky and Spandöck, Akustische Zeits. **2**, 322 (1937); see also E. Bedell, J. Acous. Soc. Am. **8**, 118 (1936).

On New Results in Research on Violins. H. BACKHAUS AND G. WEYMANN, Akustische Zeits. **4**, 302 (1939).—The desirability of being able to obtain the response of a violin as a continuous function of frequency in a short length of time has led to the development of a new measuring technique described here. Fig. 1a shows a violin response curve obtained by an older method of exciting the violin with a mechanical bowing mechanism consisting of a weighted bow constrained to move backward and forward in one dimension only. The tones produced were analyzed into their harmonic components and the curve built up by using the first harmonic of the g string and the first, second and fourth harmonics of the e string.[1] By the new

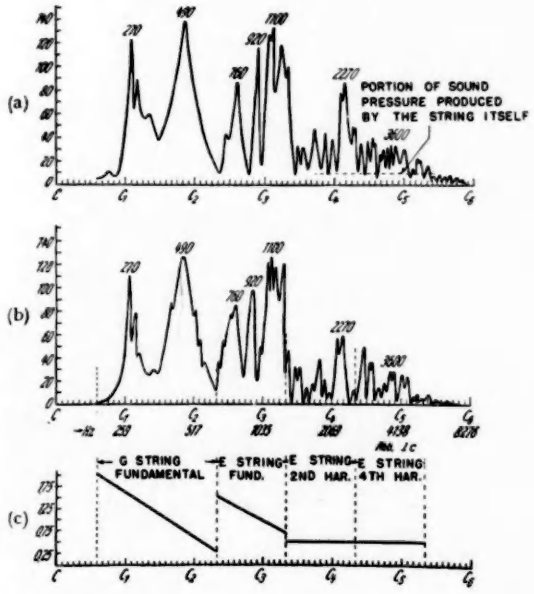

FIG. 1. Response of a violin: (a) string excitation, (b) electromagnetic excitation, (c) current amplitude in electromagnet.

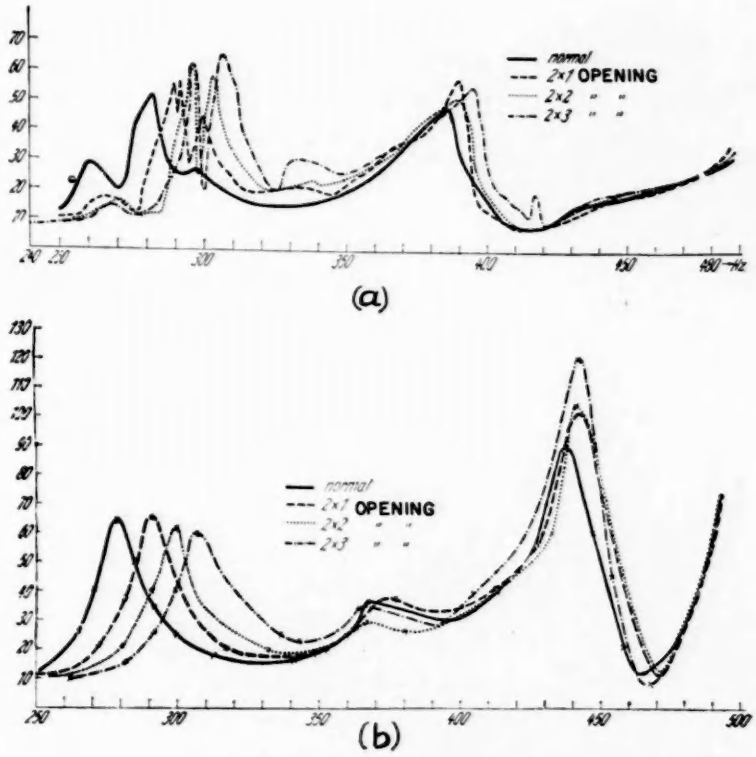

FIG. 2. Response of violin *P*-2 in the vicinity of the air chamber resonance.

method, the strings were damped so that no sound emanated from them. The armature of a special electromechanical vibrating system was pressed against the violin bridge in such a way as to excite it perpendicular to the strings and perpendicular to the surface of the violin. The authors state that the armature used had no elastic restoring force and that the mass loading on the bridge was so small as not to shift the natural frequencies appreciably.

Since the electromagnetic coil of the driving unit was largely inductive, a current inversely proportional to frequency for constant applied voltage was obtained. By using this decreasing current characteristic and readjusting the current amplitude at those frequencies at which the bow would normally shift from one string to the next, a response curve (Fig. 1b) was obtained which resembles almost exactly that of Fig. 1a. The current through the coil of the driving unit is shown in Fig. 1c. It was found necessary to maintain constant current for frequencies above about 1100 cycles to bring about agreement between the two curves. The effect of not allowing the string to radiate was chiefly observable at frequencies above 2000 cycles. Above 5000 cycles measurements showed that normally about half of the energy was radiated directly from the string. The time required for making an analysis, such as shown in Fig. 1b, was about thirty minutes.

Using this method of excitation, the authors made directionality measurements on a number of violins. They concluded that there is no specific best location for the microphone during tests. The old assumption that all violins of the ordinary shape have the same directional characteristics for both medium and low frequencies is not valid above 900 cycles. Measurements on the output of a violin for a single microphone placement were judged not to be of high significance above this frequency.

The main cavity resonance of a violin occurs at a frequency near 270 cycles. This resonance acts chiefly on the bottom; the amplitude of its vibration being roughly twice that of the top. This, perhaps, is due to the fact that the top is stiffened by the bass bar. Because the sound post rigidly joins the top and bottom together, the two are constrained to move in phase. At the *f* holes the pressure amplitudes measured were roughly ten times that of the rest of the body. Theoretically, if the *f* holes were imagined replaced by circular holes of the same area and vibrating with an amplitude ten times that of the rest of the body, then the sound pressure at a remote point would be enhanced by only 25 percent due to the presence of the holes. Therefore, the authors concluded that the largest part of the output at resonance is due to the main part of the body and that the radiation from the holes is of secondary importance.

Figure 2 shows the acoustical response curves of a violin for different sizes of *f* hole openings at frequencies in the vicinity of the main cavity resonance. Note that the frequency of the resonance moved upward as the conductance of the holes became larger. The curves in Fig. 2a were obtained with the violin clamped both at the neck and at the end of the body, while Fig. 2b was obtained for

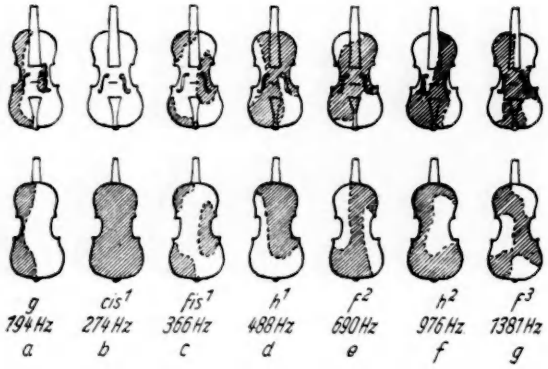

FIG. 3. Vibration patterns of a violin at specific well-defined resonant frequencies.

the violin clamped only at the neck. With neck clamping only the second resonance peak near 300 cycles did not appear. This spurious peak was attributed by the authors to a resonance in the neck.

For vibrating plates and spheres the acoustical output can be computed and depends on the surface velocity and the radiation resistance. The radiation resistance increases with lower order of source, i.e., with less nodal lines. For the violin, the lowest (zero) order of vibration is shown in Fig. 3b, corresponding to the cavity resonance discussed above. For this mode of vibration the radiation resistance is sizable. On two violins tested, a second zero-order mode of vibration occurred just below 700 cycles, with sizable output. Just above this frequency the output of most violins is partly filled in.[1] This is, perhaps, due to the existence of a favorable mode of vibration which is not present in all instruments. For certain frequencies the pressure maxima were associated with well-defined nodal lines of vibration. In other instances the nodal lines were not well defined, especially in the vicinity of resonances characterized by double peaks in the response curve. In many cases no change was found in vibration patterns over a range of several tones.

Meinel[2] described experiments wherein the body thickness was varied by scraping down the wood from the outside. In this experiment many of the vibrational patterns in Fig. 3 were preserved throughout all stages of the scraping, though deviations were more likely to occur on the bottom side than on the top. The authors state that the fact that the wood thickness was always varied from the outside places a definite restriction on the generality of these results. They conclude by saying that it *cannot* be said that: (a) all violins have the same modes of vibration, and (b) quality is independent of vibrational patterns.—L. BERANEK.

[1] H. Meinel, Akustische Zeits. **4**, 89 (1939); L. Beranek, J. Acous. Soc. Am. **11**, 259 (1939).
[2] H. Meinel, Elek. Nachr.-Techn. **14**, 119 (1937).

Construction and Properties of a Variable Acoustic Resistance Standard. K. SCHUSTER AND W. STÖHR, Akustische Zeits. **4**, 253 (1939).—There is a known demand for a variable acoustic resistance which will serve as a comparison resistance for the measurement of acoustical resistance by a bridge method.[1] One type is described in this paper.

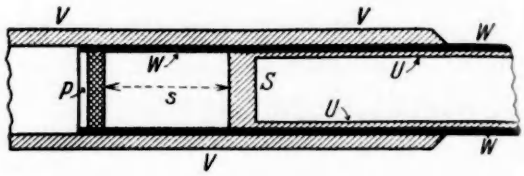

FIG. 1. Construction of variable resistance standard.

The resistance standard consisted of a disk of absorbing material P of a certain thickness (see Fig. 1), behind which there was an air space of variable length s. The absorbing disk was mounted on the end of a tube W (the "resistance tube") which in turn was inserted into the main tube V of the bridge. Inside the resistance tube there was another tube which was closed and served as a rigid piston ("piston tube"). At any fixed plane F in the main tube, the acoustical impedance can be expressed through the reflection coefficient R and a phase shift θ. A variation in the position of the resistance tube W will essentially vary only the value of θ at F. If, however, the distance s is varied, both R and θ vary at F.

For any fixed frequency the absorption coefficient, $a=1-R$, will become a maximum for some particular length s_m. If the thickness of the absorbing disk is chosen properly the absorption coefficient will become almost 100 percent. The authors call this best disk thickness "H" and derive an approximate formula for it.

$$Hr=\sigma c\rho, \tag{1}$$

where ρc is the specific resistance of air, σ is the porosity of the material, and r is the specific resistance for air flow. Investigation of the more accurate equation and experimental observations show, in the frequency region of a thousand cycles or more, and if r is not too small, that deviations in H of as much as 20 percent from the calculated value will only cause a change in maximum absorption from 100 to 99 percent.

Characteristic absorption curves for three types of absorbing materials are shown in Fig. 2. (*a*) is for a wool-felt material ($r\doteq 50$ c.g.s., $\sigma\doteq 0.8$); (*b*) is for a Celotex-type material ($r\doteq 20$ c.g.s., $\sigma\doteq 0.1$); (*c*) is for a Dyckerhoff-A acoustical material ($r\doteq 10$ c.g.s., $\sigma=0.65$).

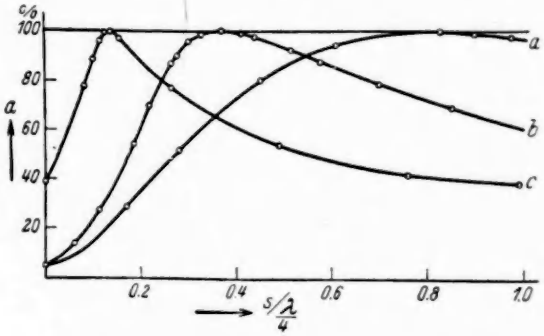

FIG. 2. Characteristic lines of three "resistance tubes."

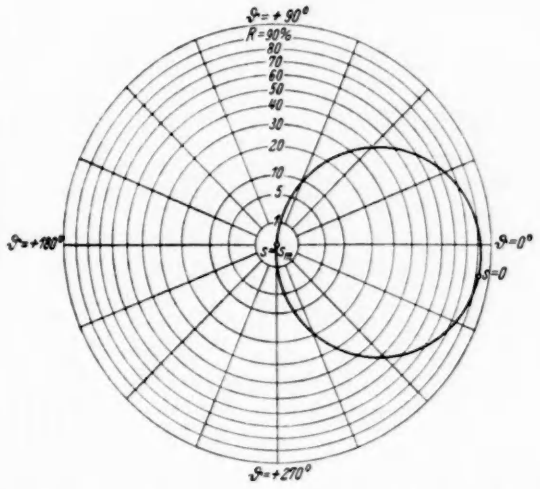

FIG. 3. Input impedance of the resistance standard with s as parameter.

In addition to the best thickness H and the length s_m for maximum absorption, still another absorption, a_0, is important in the design of a resistance standard. This absorption occurs for $s=0$, and is important in extending the measuring region to small absorptions. Examination of Fig. 2 shows that different absorbing discs of best thickness possess, in general, different a_0 values. To a first approximation the value of the reflection coefficient for $s=0$ (using the best thickness H) is given by:

$$R_0=1-\frac{4}{3}\sigma^4\cdot\frac{\omega^2\rho^2}{r^2}$$

or

$$a_0=\frac{4}{3}\sigma^4\cdot\frac{\omega^2\rho^2}{r^2}. \qquad (2)$$

This formula shows that the smaller the porosity and the greater the resistance for air flow, the smaller a_0 will be. According to (1) such a material has a proportionally small best disk thickness. However, for stability, the disk thickness should not be made smaller than one mm.

In order to calibrate the variable resistance standard, the authors chose to use the bridge arrangement itself. The method used was to compare the standard resistance being calibrated with the input resistance of a rigidly terminated tube. The input reflection coefficient and phase angle, R_1, θ_1, of a tube of length l terminated with a reflector of constants R_2, θ_2 are given by,

$$R_1=R_2e^{-4\beta l},$$
$$\theta_1=\theta_2-2\alpha l\pm 2n\pi, \qquad (3)$$

where $\alpha=\omega/c$, and β is the damping constant for the tube. Experimental evidence[2] indicates that the true β-value is about 15 percent greater than that predicted by the Kirchhoff formula. The values used in this investigation were determined by the relation:

$$\beta=3.3\cdot 10^{-5}(\sqrt{\nu}/A)\cdot(1+0.0012t), \qquad (4)$$

where ν is the frequency, A is the radius of the tube, and t is temperature in degrees C. For a rigid termination, $R_2=1$ and $\theta_2=0$.

For calibration, the variable resistance standard was placed in one arm of the bridge, and the rigidly terminated tube (calibration tube) was placed in the other. If, after balance, the distance from the center point of the bridge to the calibration tube was b and to the variable standard d, then, assuming bridge symmetry, it can be written that:

$$R=e^{-4\beta(b-d)},$$
$$\theta=-2\alpha(b-d)\pm 2n\pi. \qquad (5)$$

The number n is always chosen such that θ lies between 0° and 360°. The calibration was carried out for a number of discrete values of s.

It is apparent, due to the multiplicity of 2π in (5), that balance can be obtained for a number of certain discrete values of s and $b-d$, say s_n and $(b-d)_n$. This fact can be shown graphically by plotting on the impedance plane, first, a curve of input impedance of the variable standard for a variation of s, and, second, a curve of input impedance of a rigidly terminated tube for variation of l. The impedance plane chosen was not the conventional one with the real and imaginary parts as Cartesian coordinates, but with $\sqrt{R}$ as a radius vector and θ as a polar angle (Fig. 3). The impedance curve of the variable standard (plotted) is roughly a circle which passes through the origin for $s=s_m$ and approaches closest to the arc $\sqrt{R}=1$ at $s=0$. The impedance curve of the calibration tube is a spiral about the origin, whose revolutions become closer, the smaller β is. Balances are possible only at points of intersection of the curves, and the s values corresponding to these points are the values s_n.

Equation (3) would indicate that for $R_2=1$, l would have to become very long as R approaches zero. To avoid the use of excessively long tubes in calibrating the variable standard in the vicinity of unity absorption, a method was adopted by which two variable standards were calibrated simultaneously. Call the standards I and II. First I was placed in one arm of the bridge and a short calibration tube placed in the other arm. A calibration point for I was obtained at a length $s=s_1$. Then I was substituted in place of the rigid termination of the calibration tube and II was put in the other arm of the bridge and balance repeated. Note that for this latter case R_2 in (3) was no longer unity. By placing I and II alternately in place of the rigid termination, calibration of the standards was carried out over the entire range of variation of s.

One further difficulty arose. Inspection of Fig. 3 shows that in the vicinity of $s=0$, variations in s do not essentially change the reflection coefficient, but only cause a variation in the phase angle. Hence, when making balances in this region, a number of balances were obtained for values of s and d such that the sum $(s+d)$ was essentially a constant. This was true because variation of d causes only a variation in the phase angle. To avoid this difficulty the following procedure was adopted: First, the best balance was obtained by varying the position of the resistance tube, keeping the piston tube in a fixed position with respect to the main tube. This was accomplished by binding the piston tube rigidly to the main tube with the aid of a rod. Then the balance was completed by varying either the

piston tube alone, or by varying the entire assembly with respect to the main tube. In the first part of the balance only the value of R is varied. In the second part only the value of θ is varied. This method of balancing arrived at answers free from error. To measure small absorptions, variable standards can be built having absorbing discs with thicknesses other than the "best" values, H. In this case no variable rod would be necessary.

Regarding stability over long periods of time, one standard varied by 4 percent over a period of six months. This was thought to be caused by condensation of oil in the pores of the material which resulted in a change in its porosity.—L. BERANEK.

[1] K. Schuster, Elek. Nachr.-Techn. **13**, 164 (1936); E. Waetzmann, Akustische Zeits. **3**, 1 (1938); N. Robinson, Phil. Mag. **23**, 665 (1937).
[2] Waetzmann and Keibs, Ann. d. Physik (5), **22**, 247 (1935); Oberst, Akustische Zeits. **2**, 76 (1937).

APRIL, 1940 J. A. S. A. VOLUME 11

References to Contemporary Papers on Acoustics

MOST of the titles of papers appearing in other languages than English have been translated; Abstracts in English of many foreign papers have appeared or soon will appear in Science Abstracts, Section A. Where references are made to Science Abstracts, the reference is to the volume number and abstract number. The abbreviations of the names of journals are those used in Science Abstracts and can be found in any annual index to those abstracts.

The numbers appearing at the head of each entry are the numbers which identify the subject headings in the Cumulative Index of this journal issued in November, 1939. This same Classification of Subjects can be found in the index to Volume 11.

Compiled with the generous assistance of F. E. Orton, Physics Librarian, University of Michigan.

2. ARCHITECTURAL ACOUSTICS

2.1 **Control of Sound in Buildings.** Architectural Record **87**, 66–73 (1940).

2.1 **Architectural Acoustics; Sound, Its Fundamental Properties and Behavior.** S. L. MACDONALD AND P. M. MORSE. Architectural Forum **71**, 131–134 (1939).

2.2 **Community Theaters: Acoustic Requirements.** H. BURIS-MEYER AND E. C. COLE. Architectural Record **86**, 94 (1939).

2.2 **Acoustic Condition Factors (in a room or studio).** M. RETTINGER. J. Soc. Mot. Pict. Eng. **33**, 410–420 (Oct. 1939).

2.2 **Controlled Sound Reflection in Review Rooms, Theatres, etc.** C. M. MUGLER. J. Soc. Mot. Pict. Eng. **33**, 421–425 (Oct. 1939).

2.4 **Eine Methode zur Bestimmung des Akustischen Reflexionsvermögens von Bodenflächen.** (A Method of Determining the Acoustical Reflectivity of the Earth's Surface.) W. ERNSTHAUSEN AND W. v. WITTERN. Akustische Zeits. **4**, 353–359 (Nov. 1939).

2.5 **Eine Methode zur Bestimmung des Akustischen Reflexionsvermögens von Bodenflächen.** (A Method of Determining the Acoustical Reflectivity of the Earth's Surface.) W. ERNSTHAUSEN AND W. v. WITTERN. Akustische Zeits. **4**, 353–359 (Nov. 1939).

2.7 **Synthetic Reverberation.** P. C. GOLDMARK AND P. S. HENDRICKS. J. Soc. Mot. Pict. Eng. **33**, 635–649 (Dec. 1939). Also J. I. R. E. **27**, 747–752 (Dec. 1939).

2.7 **Synthetic Reverberation; Recording Fugitive Sound Pattern of Original Program Signal on Rotating Phosphor-Coated Disk by Means of Modulated Light Source and Simple Optical System.** P. C. GOLDMARK AND P. S. HENDRICKS. I. R. E., Proc. **27**, 747–752 (1939).

2.7 **Acoustical Control by Acousti-Celotex Installations.** Gr. Britain and the East **52**, Supp. 16–17 (April 6, 1939).

2.7 **An Arrangement for Regulating the Acoustic Properties of Rooms.** W. OELSNER. Hochfrequenztechn. u. Electroakustik **54**, 105 (1939).

2.7 **Das Schallspektrum von Knallfunken und Knallpistolen mit einem Beitrag über Anwendungsmoglichkeiten in Elektroakutischen Messtechnik.** (The Sound Spectrum of Spark and Piston Noises, with a Suggestion Concerning Its Possibilities of Application in Electroacoustical Measuring Technique.) W. WEBER. Akustische Zeits. **4**, 373–390 (Nov. 1939).

2.8 **Das Schallspektrum von Knallfunken und Knallpistolen mit einem Beitrag über Anwendungsmoglichkeiten in Elektroakutischen Messtechnik.** (The Sound Spectrum of Spark and Pistol Noises, with a Suggestion Concerning Its Possibilities of Application in Electroacoustical Measuring Technique.) W. WEBER. Akustische Zeits. **4**, 373–390 (Nov. 1939).

2.9 **National Physical Laboratory; Measurement of Sound, Noise Abatement, Sound Transmission through Walls, Floors and Pipes.** Engineering **148**, 494–495 (1939).

2.9 **Vorschlag zur Definition der Trittschalldämmung.** (Proposed Definition of Foot-fall Sound Attenuation.) P. HALLER. Akustische Zeits. **4**, 370–372 (Nov. 1939).

2.9 **Structural Requirements of Houses: Insulation; Thermal and Acoustical.** Compiled by W. H. HAYES. Architectural Record **86**, 85–86 (1939).

2.9 **Transmission of Sound through Thin Plates.** F. H. SANDERS. Canad. J. of Research **17**, 179–193 (1939). Sci. Abs. **A42**, 4372 (1939).

3. BOOKS AND BIBLIOGRAPHIES

3.1 **Leitfaden zur Berechnung von Schallvorgängen.** (Charts for Computing Sound Radiation.) HEINRICH STENZEL. (Julius Springer, Berlin), book of 124 pages, 106 figures.

3.1 **Einführung in die Akustik.** FERDINAND TRENDELENBERG. (Julius Springer, Berlin), book of 277 pages.

3.2 **Bibliography on Phonetics.** S. N. TREVINS. American Speech. (Oct. 1939, and Dec. 1939.)

4. EAR AND HEARING

4.1 **Auditory Patterns.** H. FLETCHER. Rev. Modern Physics **12**, 47–65 (1940).

4.1 **Hören in der Stille.** (Hearing in Stillness.) E. WAETZMANN. Nachrichten von der Gesellschaft der Wissenschaften zu Göttingen (2) **3**, 1–30 (1939).

4.4 **Analysis of World's Fairs' Hearing Tests.** H. C. MONTGOMERY. Bell Lab. Record **18**, 98–103 (Dec. 1939).

4.5 **Synthephone.** C. A. CULVER AND D. T. WILLIAMS. Rev. Sci. Instruments **10**, 319–21 (1939).

4.5 **An Artificial Mastoid for Audiphone Measurements.** M. S. HAWLEY. Bell Lab. Record **18**, 73–75 (Nov. 1939).

4.6 **Über die Empfindlichkeit des Stehenden und Sitzenden Menschen Gegen Sinusformige Erschütterungen.** (The Sensitivity of Standing and Sitting Observers to Sinusoidal Vibrations.) GEORG V. BÉKÉSY. Akustische Zeits. **4**, 360–369 (Nov. 1939).

4.6 **Quantitative Hörfahigkeitsprüfurgen an Schülern.** (Quantitative Examination of the Hearing Capacity of Students.) P. EICHLER. Unterichtsblätter für Mathematik und Naturwiss. **45**, 99–103 (1939).

4.6 **Analysis of World's Fairs' Hearing Tests.** H. C. MONTGOMERY. Bell Lab. Record **18**, 98–103 (Dec. 1939).

4.9 **Thyratron Inflection Indicator for Teaching the Deaf.** T. A. STERNE AND H. J. ZIMMERMANN. Journ. Sci. Instruments **16**, 334–336 (1939). Sci. Abs. **A42**, 4371 (1939).

4.10 **Combination Tones in Sound and Light.** W. BRAGG. Nature **143**, 542–545 (1939).

4.10 **Frequency Produced by the Combination of Two Vibrations of Nearly Equal Frequency.** G. E. OWEN. Am. Phys. Teacher **7**, 177–178 (1939).

4.10 **Hören in der Stille.** (Hearing in Stillness.) E. WAETZMANN. Nachrichten von der Gesellschaft der Wissenschaften zu Göttingen (2) **3**, 1–30 (1939).

5. APPLIED ACOUSTICS, INSTRUMENTS AND APPARATUS

5.3 **Das Schallspektrum von Knallfunken und Knallpistolen mit Einem Beitrag über Anwendungsmoglichkeiten in Elektroakutischen Messtechnik.** (The Sound Spectrum of Spark and Pistol Noises, with a Suggestion Concerning Its Possibilities of Application in Electroacoustical Measuring Technique.) W. WEBER. Akustische Zeits. **4**, 373–390 (Nov. 1939).

5.4 **Automatic Cathode-Ray Oscillograph.** W. L. GAINES. Bell Lab. Record **18**, 145–150 (Jan. 1940).

5.4 **Recording Transient Disturbances.** O. D. GRISMORE. Bell Lab. Record **18**, 140–144 (Jan. 1940).

5.4 **Analysis of Varying Sound.** K. IMAHOVI. Nature **144**, 708 (1939).

5.4 **The Filtration of Sound in Non-homogeneous Media.** R. B. LINDSAY. Science **90**, 409 (1939).

5.7 **Thyratron Inflection Indicator for Teaching the Deaf.** T. A. STERNE AND H. J. ZIMMERMANN. Journ. Sci. Instruments **16**, 334–336 (1939). Sci. Abs. **A42**, 4371 (1939).

5.9 **The Measurement of Pressure, Movement, Acceleration and Other Mechanical Quantities, by Electrostatic Systems.** C. H. W. BROOKES-SMITH AND J. A. COLLS. Journ. Sci. Instruments **16**, 361–365 (Dec. 1939).

5.9 **Dynamisches Tauchspulen-Mikrophon der AEG.** (Dynamic Moving Coil Microphone of AEG.) H. LAUFFER. Hochfrequenztechn. u. Elektroakustik **54**, 134–136 (Oct. 1939).

5.10 **Ears under Water; Anti-Submarine Warfare.** Time **34**, 66 (Oct. 9, 1939).

5.10 **Listening on War; Microphones Find Both Submarines and Planes.** F. THONE. Science News Letter **36**, 246–247 (1939).

5.11 **Bridge Oscillator.** Bell Lab. Record **18**, 144 (Jan. 1940).

5.11 **Beat Notes in High-Frequency Calibration.** F. R. STANSEL. Bell Lab. Record **18**, 116–118 (Dec. 1939).

5.15 **Computation of the Composite Noise Resulting from Random Variable Sources; Statistical Method for Computing Reading on a Sound Level Meter.** E. DIETZE AND W. D. GOODALE. Bell System Techn. J. **18**, 605–623 (1939).

5.15 **Acoustic Meter.** V. FEDOROVICH AND S. SALTIKOV. J. Techn. Phys. **9**, 737–742 (1939).

5.15 **Automatic Cathode-Ray Oscillograph.** W. L. GAINES. Bell Lab. Record **18**, 145–150 (Jan. 1940).

5.15 **Recording Transient Disturbances.** O. D. GRISMORE. Bell Lab. Record **18**, 140–144 (Jan. 1940).

5.15 **Measurement of Sound Intensity under Water by Resonant Rayleigh Disc.** T. HAYASHI. E. T. J. **3**, 175–178 (1939). Sci. Abs. **B42**, 2347 (1939).

5.16 **Report on the Adaptation of Fine-Grain Films to Variable Density Sound Technics.** J. Soc. Mot. Pict. Eng. **34**, 3–11 (Jan. 1940).

5.16 **Sound Picture Standards.** J. Soc. Mot. Pict. Eng. **34**, 88–93 (Jan. 1940).

5.16 **A New Magnetic Recorder and Its Adaptations.** S. J. BEGUN. J. Soc. Mot. Pict. Eng. **33**, 538–546 (Nov. 1939).

5.16 **Class B Push-Pull Recording for Original Negatives.** D. J. BLOOMBERG AND C. L. LOOTENS. J. Soc. Mot. Pict. Eng. **33**, 664–669 (Dec. 1939).

5.16 **New 16-mm. Recording Equipment.** D. CANADY. J. Soc. Mot. Pict. Eng. **33**, 571–572 (Nov. 1939).

5.16 **Notes on French 16 mm.** (recording and reproducing) **Equipment.** D. CANADY. J. Soc. Mot. Pict. Eng. **33**, 573–577 (Nov. 1939).

5.16 **A Densitometric Method of Checking the Quality of Variable-Area Prints.** C. R. DAILY AND I. M. CHAMBERS. J. Soc. Mot. Pict. Eng. **33**, 398–402 (Oct. 1939).

5.16 **Improvement in Sound and Picture Release through the Use of Fine-Grain Film.** C. R. DAILY. J. Soc. Mot. Pict. Eng. **34**, 12–25 (Jan. 1940).

5.16 **A Direct Positive System of Sound Recording** (on film). DIMMICK AND BLANEY. J. Soc. Mot. Pict. Eng. **33**, 479–487 (Nov. 1939).

5.16 **Optical Control of Wave-Shape and Amplitude Characteristics in Variable-Density Recording.** G. L. DIMMICK. J. Soc. Mot. Pict. Eng. **33**, 650–663 (Dec. 1939).

5.16 **Effect of Orientation of the Scanning Image on Quality of Sound Reproduced from Variable-Width Records.** D. FOSTER. J. Soc. Mot. Pict. Eng. **33**, 502–516 (Nov. 1939).

5.16 **Synthetic Reverberation.** P. C. GOLDMARK AND P. S. HENDRICKS. J. Soc. Mot. Pict. Eng. **33**, 635–649 (Dec. 1939). Also J. I. R. E. **27**, 747–752 (Dec. 1939).

5.16 **Synthetic Reverberation; Recording Fugitive Sound Pattern of Original Program Signal on Rotating Phosphor-Coated Disk by Means of Modulated Light Source and Simple Optical System.** P. C. GOLDMARK AND P. S. HENDRICKS. I. R. E., Proc. **27**, 747–752 (1939).

5.16 **A Light-Weight Sound-Recording System.** HOPPER MANDERFELD, AND SCOVILLE. J. Soc. Mot. Pict. Eng. **33** 449–457 (Oct. 1939).

5.16 **A New Mobile Film-Recording System.** B. KREUZER AND C. L. LOOTENS. J. Soc. Mot. Pict. Eng. **33**, 382–393 (Oct. 1939).

5.16 **Controlled Sound Reflection in Review Rooms, Theatres, etc.** C. M. MUGLER. J. Soc. Mot. Pict. Eng. **33**, 421–425 (Oct. 1939).

5.16 **Modern Instantaneous Recording and Its Reproduction.** NEELY AND STANCIL. J. Soc. Mot. Pict. Eng. **33**, 547–550 (Nov. 1939).

5.16 **Volume Distortion.** (Sound reproduced at different level than original.) S. L. REICHES. J. Soc. Mot. Pict. Eng. **34**, 59–75 (Jan. 1940).

5.16 **Acoustic Condition Factors** (in a room or studio). M. RETTINGER. J. Soc. Mot. Pict. Eng. **33**, 410–420 (Oct. 1939).

5.16 **A Sound-Track Center-Line Measuring Device.** F. W. ROBERTS AND H. R. COOK, JR. J. Soc. Mot. Pict. Eng. **34**, 38–42 (Jan. 1940).

5.16 **Photographic Duping of Variable-Area Sound.** F. W. ROBERTS AND E. TAENZER. J. Soc. Mot. Pict. Eng. **34**, 26–37 (Jan. 1940).

5.16 **A New Classroom 16-mm. Sound Projector** (Ampro). A. SHAPIRO. J. Soc. Mot. Pict. Eng. **33**, 695 (Dec. 1939).

5.16 **The Polyrhetor—A 150-Channel Film Reproducer.** STANTON, MARION, AND WATERS. J. Soc. Mot. Pict. Eng. **33**, 488–501 (Nov. 1939).

5.17 **Theory of the Telephone Receiver.** A. FERRARI-TONIOLO. I. E. N. G. F. No. 51, pp. 23 (1939). Sci. Abs. **B42**, 2425 (1939).

5.17 **Gas-Tube Noise Generator for Circuit Testing.** E. PETERSON. Bell Lab. Record **18**, 81–83 (Nov. 1939).

5.51 **Synthetic Reverberation; Recording Fugitive Sound Pattern of Original Program Signal on Rotating Phosphor-Coated Disk by Means of Modulated Light Source and Simple Optical System.** P. C. GOLDMARK AND P. S. HENDRICKS. I. R. E., Proc. **27**, 747–752 (1939).

5.51 **Measuring Lines for Program Transmission.** S. J. HARAZIM AND J. M. HUDACK. Bell Lab. Record **18**, 108–112 (Dec. 1939).

5.51 **Acoustic Treatment and Measurements of the New Bombay Studios of All-India Radio.** N. L. SAHDEV. Electrotechnics, Bangalore No. 12, 64–76 (Aug. 1939).

6. MUSICAL INSTRUMENTS AND MUSIC

6.1 **Our Insect Musicians.** L. E. DUNKIN. Etude **57**, 438 (1939).

6.1 **Approximate Values for the Calculation of Tempered Note Frequencies, and Their Application to the Drive of Fully Electric Organs.** F. TRAUTWEIN. Akustische Zeits. **4**, 261–262 (1939).

6.5 **Les Propriétés Musicales des Jets Fluids.** (Musical Properties of Fluid Jets.) H. VIGNERON. La Nature **67**, Part 1, 49–51 (1939). Reply: Mourey **67**, Part 1, 191 (1939).

6.6 **A Note on the Effect of Finite Breadth of the Hammer Striking a Pianoforte String.** D. BASU. Current Science **8**, 109–110 (1939).

6.6 **Notes on the General Theory of Impact of an Elastic Hammer on a Finite Pianoforte String.** GHOSH. Science and Culture, Calcutta **5**, 132–133 (1939).

6.6 **Theory and Practice of the Excitation of Piano Strings.** I. N. JAKOVLEV. J. Techn. Phys. USSR 9, 312–328 (1939) (In Russian). Sci. Abs. **A42**, 4640 (1939).

6.6 **Approximate Values for the Calculation of Tempered Note Frequencies, and Their Application to the Drive of Fully Electric Organs.** F. TRAUTWEIN. Akustische Zeits. **4**, 261–262 (1939).

6.7 **A Sciertific Search for the Secret of Stradivarius.** F. A. SAUNDERS. Frank. Inst., J. **229**, 1–20 (1940).

6.8 **The Stability of the Sound Emitted by a Tube with a Beating Reed under a Constant Pressure.** L. AUGER. Comptes Rendus **209**, 508–509 (1939).

6.9 **A New Electrical Musical Instrument: the "Stoytone."** Scientific American **161**, 294–295 (1939).

6.9 **Synthephone.** C. A. CULVER AND D. T. WILLIAMS. Rev. Sci. Instruments **10**, 319–21 (1939).

6.9 **Electronic Church Chimes.** F. Dostal. Electronics **12**, No. 8, 18–19 (Aug. 1939).

6.9 **Electrical Musical Instruments (Survey).** A. Fervari-Toniolo. Boll. del Centro Volpi di Elett., English Edition **1**, 95 (1938).

7. Noise

7.1 **To Combat Noise; Westinghouse Devices for Study of Sound.** J. A. Baubie. Scientific American **162**, 10–11 (1940).

7.1 **Über die Empfindlichkeit des Stehenden und Sitzenden Menschen Gegen Sinusformige Erschütterungen.** (The Sensitivity of Standing and Sitting Observers to Sinusoidal Vibrations.) Georg v. Békésy. Akustische Zeits. **4**, 360–369 (Nov. 1939).

7.1 **Noise Control.** P. H. Smith. Scientific American **161**, 204–207 (1939).

7.1 **Gas-Tube Noise Generator for Circuit Testing.** E. Peterson. Bell Lab. Record **18**, 81–83 (Nov. 1939).

7.5 **Das Schallspektrum von Knallfunken und Knallpistolen mit Einem Beitrag über Anwendungsmoglichkeiten in Elektroakutischen Messtechnik.** (The Sound Spectrum of Spark and Pistol Noises, with a Suggestion Concerning Its Possibilities of Application in Electroacoustical Measuring Technique.) W. Weber. Akustische Zeits. **4**, 373–390 (Nov. 1939).

8. Standards

8.1 **Sound Picture Standards.** J. Soc. Mot. Pict. Eng. **34**, 88–93 (Jan. 1940).

9. Speech and Singing

9.1 **Starting Characteristics of Speech Sounds.** R. O. Drew and C. W. Kellogg. J. Soc. Mot. Pict. Eng. **34**, 43–58 (Jan. 1940).

9.1 **New Approaches to the Science of Voice.** C. E. Seashore. Scientific Monthly **49**, 340–350 (1939).

9.1 **Bibliography on Phonetics.** S. N. Trevins. American Speech. (Oct. 1939 and Dec. 1939.)

9.2 **The Physiology of the Baritone Voice.** R. Curry. Radiologica **2**, 211–216 (1938).

9.5 **Analysis of a Perceptible Series of Partials in Vocal Sound.** D. Lewis and W. H. Lichte. J. of Experimental Psychology **24**, 254–267 (1939).

9.6 **Sonovox.** Time **34**, 46 (July 24, 1939).

9.6 **Automatic Synthesis of Speech.** H. Dudley. Nat. Acad. Sci., Proc. **25**, 377–383 (1939).

9.6 **The Vocoder.** Homer Dudley. Bell Lab. Record **18**, 122–126 (Dec. 1939).

9.6 **Gas Tube Noise Generator for Circuit Testing.** E. Peterson. Bell Lab. Record **18**, 81–83 (Nov. 1939).

9.6 **The Polyrhetor—A 150-Channel Film Reproducer.** Stanton, Marion, and Waters. J. Soc. Mot. Pict. Eng. **33**, 488–501 (Nov. 1939).

10. Supersonics (Ultrasonics)

10.1 **Some New Remarks on the Supersonic Analogy of the Electromagnetic Field.** D. Riabouchinsky. Comptes Rendus **209**, 587–581 (1939).

10.2 **Sonoluminescence and Sonic Chemibuminescence.** E. N. Harvey. Am. Chem. Soc., J. **61**, 2392–2398 (1939). Sci. Abs. **A42**, 4642 (1939).

10.2 **The Influence of the Angle of Incidence of Light on the Diffraction of Light by Supersonic Waves.** E. A. Neumann. Phys. Soc., Proc. **51**, 794–802 (1939).

10.2 **Visible Stationary Supersonic Waves in Liquids.** O. Nomoto. Phys. Math. Soc. Japan, Proc. **21**, 493–496 (1939). (In German.) Sci. Abs. **A42**, 4643 (1939).

10.2 **Über die Möglichkeit Einer Akustischen Abbildung in Analogie zur Optischen.** (Possibility of Acoustic Images Analogous to Optical.) R. Pohlmann. Zeits. f. Physik **113**, 697–709 (1939).

10.2 **Supersonic Dispersion and Infra-Red Radiation.** E. G. Richardson, Nature **143**, 638–639 (1939).

10.3 **Ultrasonic Stroboscope for Measuring Sound Wave-Length in Liquids.** F. E. Fox and G. D. Rock. Rev. Sci. Instruments **10**, 345–358 (1939).

10.3 **Fixed Path Acoustic Interferometer for the Study of Matter.** J. C. Hubbard and I. F. Zartman. Rev. Sci. Instruments **10**, 382–386 (1939).

10.4 **Supersonic Effects of Water Vapor in CO_2 and Their Relation to Molecular Vibrations.** W. H. Peilemeier *et al.* J. Chem. Phys. **8**, 106–15 (1940).

10.4 **Über Hörschallgeschwindigkeit und Dispersion in Chlor.** (Sound Velocity and Dispersion in Chlorine.) R. Schulze. Ann. d. Physik (5) **34**, 41–59 (1939).

10.5 **The Dispersion of Ultrasonic Waves in Liquids.** R. Bär. *Chandrasekhara Venkata Raman, Jubilee Volume* (1938), pp. 289–293.

10.5 **Dispersion and Selective Absorption in the Propagation of Ultrasound in Fluids Contained in Tubes.** G. S. Field. Canad. J. of Research **A17**, 197–201 (1939).

10.5 **Velocity of Ultrasound in Liquids as a Function of the Temperature.** P. Prozorov and V. Nozdrev. J. Experimental and Theoretical Physics (USSR) **9**, 625–629 (1939).

10.5 **Absorption of Supersonics in Mercury.** P. Rieckmann. Phys. Zeits. **40**, 582–590 (1939).

10.5 **Determination of the Molecular Radii of Organic Liquids from Velocity of Sound and Density.** W. Schaafs. Zeits. f. Physik **114**, 110–115 (1939).

10.5 **La Dispersion des Ondes Ultrasonores Dans l'huile de Vicin.** (The Dispersion of Supersonic Waves in Castor Oil.) L. Zachoval. Comptes Rendus **208**, 265–266 (1939).

10.5 **Dispersion of Supersonic Waves in Liquids.** L. Zachoval. J. de Physique et le Radium **10**, 350–354 (1939). Sci. Abs. **A42**, 4369 (1939).

10.6 **Absorption of Sound at Solid Boundaries.** B. Konstantinov. J. Techn. Phys. (USSR) **9**, 226–238 (1939).

10.7 **Der Einfluss eines Ultraschall Feldes auf den Koagulationsgrad von Negatinen AgI-Solen durch Elektrolyte.** (The Influence of a Supersonic Field upon the Degree of Coagulation of Negative AgI Sols by Electrolytes.) J. J. HERMANS. Recueil des Travaux Chimignes des Pays-Bas **58**, 139–63 (1939).

10.7 **Orthokinetische Koagulation, die durch Schwingungen Vermsacht ist.** (Orthokinetic Coagulation which is Produced by Vibration.) J. J. HERMANS. Recueil des Travaux Chimignes des Pays-Bas **58**, 164–173 (1939).

10.7 **Über die Möglichkeit einer Akustischen Abbildung in Analogie zur Optischen.** (Possibility of Acoustic Images Analogous to Optical.) R. POHLMANN. Zeits. f. Physik **113**, 697–709 (1939).

10.7 **Some New Remarks on the Supersonic Analogy of the Electromagnetic Field.** D. RIABOUCHINSKY. Comptes Rendus **209**, 587–589 (1939).

10.7 **The Effect of Supersonic Waves on Colloidal Phenomena. V. Effect upon Thixotropie. 1. Investigations with Thixotrope Systems of Aluminum Hydroxide.** N. SATA AND N. NARUSE. Kolloid Zeits. **86**, 102–105 (1939).

10.7 **The Effect of Supersonic Waves on Colloidal Phenomena. VI. The Effect of Supersonic Waves on the Rotation of High-Molecular Colloidal Substances. 2. Experiments with Tannin Solutions.** N. SATA. Kolloid Zeits. **88**, 182 (1939).

10.7 **Influence of Supersonic Vibrations on Rotatory Power of High-Molecular Substances. Part II. Tannin Solutions.** N. SATA. Kolloid Zeits. **88**, 182–184 (1939). Sci. Abs. **A42**, 4370 (1939).

10.7 **Lowering of the Flashpoint of Mineral Oil by Supersonic Waves.** G. SADAKIYO AND T. SIRAISI. Electrot. Laborat. Tokyo, Japan, Researches No. 430 (1939), pp. 33. Sci. Abs. **B42**, 2275 (1939).

10.7 **Dispersion of Metals in Liquids in a Supersonic Field.** L. M. SOLV'EVA. Colloid J. (USSR) **5**, 289–297 (1939).

10.9 **Effect of Ultrasonic Vibration on Vaccinia Virus.** F. L. HOPWOOD *et al.* Nature **144**, 377 (1939).

10.9 **Über die Absorption des Ultraschalls im Menschlichen Gewebe und Ihre Abhandigkeit von der Frequenz.** (The Absorption of Supersonic Waves in Human Tissue and Its Dependence upon the Frequency.) R. POHLMAN. Phys. Zeits. **40**, 159–161 (1939).

11. WAVES AND VIBRATIONS

11.3 **Formula for the Change of Velocity of Sound with Temperature.** A. E. BATE. Nature **144**, 479 (1939).

11.3 **Parachor und die Schallgeschwindigkeit in den Metallischen Elementen.** (Parachor and the Sound Velocity in the Metallic Elements.) BINAYENDRA. Math. Sen. Gazzetta Chimica Italiana **68**, 662–664 (1938).

11.3 **Determination of the Velocity of Sound in a Gas: Application to Analysis of Mixtures of Helium, Oxygen, and Nitrogen.** W. B. DUBLIN *et al.* Science **90**, 399–400 (1939).

11.3 **Die Wahre Schallgeschwindigkeit in Luft.** (The True Velocity of Sound in Air.) H. O. KNESER. Ann. d. Physik (5) **34**, 665–668 (1939).

11.3 **Measurement of Sound Velocity in Liquid Argon.** H. W. LIEPMANN. Helv. Phys. Acta **12**, 421–442 (1939).

11.4 **Leitfaden zur Berechnung von Schallvorgängen.** (Charts for Computing Sound Radiation.) HEINRICH STENZEL. (Julius Springer, Berlin), book of 124 pages, 106 figures.

11.5 **The Theory of Acoustic Radiation Pressure.** C. SCHAEFER. Ann. d. Physik **35**, 473–491 (1939).

11.5 **Neue Engebrisse bei der Ausbreitung von Knallwellen in Flüssigkeiten und Festen Korpern.** (New Results in the Diffusion of Detonation Waves in Liquids and Solids.) O. v. SCHMIDT. Verhandlungen der Deutschen Physikalischen Gesellschaft. **19**, 127 (1938).

11.7 **Filtration of Sound, II.** R. B. LINDSAY. J. of Applied Physics **10**, 680–687 (1939).

11.7 **The Filtration of Sound in Non-homogeneous Media.** R. B. LINDSAY. Science **90**, 409 (1939).

11.7 **Frequency Produced by the Combination of Two Vibrations of Nearly Equal Frequency.** G. E. OWEN. Am. Phys. Teacher **7**, 177–178 (1939).

11.7 **Beat Notes in High-Frequency Calibration.** F. R. STANSEL. Bell Lab. Record **18**, 116–118 (Dec. 1939).

11.7 **Les Propriétés Musicales des Jets Fluides.** (Musical Properties of Fluid Jets.) H. VIGNERON. La Nature **67**, Part 1, 49–51 (1939). Reply: MOUREY, *ibid.* **67**, Part 1, 191 (1939).

12. GENERAL, UNCLASSIFIED

12 **Freak Effects of Sound Tested in Odd Laboratory.** A. R. BOONE. Popular Science **135**, 120–123 (1939).

12 **Answer, Echo, Answer.** A. CARR. Scientific American **161**, 342–344 (1939).

12 **Golden Age of Sound.** V. O. KNUDSEN. Popular Mechanics **72**, 828–30 (1939).

Index to Volume 11

How to Use This Index

THIS Index is of similar form to the Cumulative Index of this journal, published in November 1939, and is composed of two main divisions: The Author Index and the Subject Index.

The Subject Index to Volume 11 of the *Journal of the Acoustical Society* lists the titles of all the papers and abstracts under twelve section headings as follows: 1, Acoustical Society of America; 2, Architectural Acoustics; 3, Books and Bibliographies; 4, Ear and Hearing; 5, Instruments and Apparatus. Applied Acoustics; 6, Musical Instruments and Music; 7, Noise; 8, Standards; 9, Speech and Singing; 10, Supersonics; 11, Waves and Vibrations; 12, General, Unclassified. Each of these sections is further divided into a considerable number of decimally numbered subjects, which are listed below under the heading, "Classification of Subjects." Each subject is divided into three parts entitled Papers, Subtitles, and Abstracts. Under Papers will be found the original titles of all the papers appearing in the journal on that subject. Under Abstracts are listed the titles of the papers presented at meetings. Since a great amount of valuable material appears in the journal in papers whose titles cannot fully describe all the subjects which the papers cover, many of the subtitles of papers have been indexed under the heading Subtitles. When subtitles covering the material did not appear in the paper, they were written; the captions of some of the figures were also included under this heading. Thus, while the Subtitles portion of the Index is informal, it is hoped that it will increase the probability of finding the desired information quickly.

References on most subjects can be located immediately by a logical use of the Classification of Subjects below. In case the reference sought is of an unusual character such that there is uncertainty as to the subject under which it might have been classified, this can usually be found quite quickly by the process of eliminating the sections and subjects under which it is quite improbable that it might have been listed. In case of doubt, a title has often been listed under a number of subjects. As an example of the direct approach, suppose you wish to find data on the impedance of wool felt. On looking over the section headings it is seen that this comes under the heading Architectural Acoustics, and Subject, Measured Sound Absorption Coefficients. Under Subtitles we find a reference to wool felt and on looking up this paper we find it entitled "Effect of an Absorbing Wall on the Decay of Normal Frequencies," which title of itself would not have suggested the presence of data on wool felt, thus demonstrating the value of having the index of Subtitles.

Obviously, the Author Index is more convenient than the Subject Index when one knows the name of the author of the paper in which he is interested.

The References to Contemporary Papers on Acoustics are identified in each issue by the same decimal numbers listed below, and may be consulted for references in other journals than our own. In Volume 11 these references will be found on pages 156, 263, 370, and 494.

Classification of Subjects

	Page in this Index
1. Acoustical Society of America	
1.1 General, Unclassified	504
1.2 Constitution and By-Laws	504
1.3 History	504
1.5 Meetings	504
1.6 Members and Membership Lists	504
1.7 Letters to the Editor	504
2. Architectural Acoustics	
2.1 General, Unclassified	504
2.2 Auditorium Design (see also 7.6)	504
2.3 Studies of Existing Auditoria	505
2.4 Sound Absorption, Theory and Methods of Measurement (see also 2.5)	505
2.5 Measured Sound Absorption Coefficients	505
2.7 Reverberation, Theory and Calculation of (see also 2.3, 2.4, 2.8, 2.9 and 2.10)	505
2.9 Sound Transmission, Theory and Methods of Measurement (see also 2.9)	506
2.10 Measurements of Sound Transmission	506
2.11 Vibration Insulating Supports	506
3. Books and Bibliographies	
3.1 Book Reviews	506
3.2 Bibliographies	506

4. Ear and Hearing

4.1 General, Unclassified 506
4.2 Anatomy and Physiology of the Ear 506
4.4 Deafness (see also 5.5) 506
4.5 Instruments Relating to Hearing (see also 5.5) 507
4.6 Loudness. Threshold Determinations 507
4.7 Masking (see also 5.5) 507
4.9 Pitch 507
4.10 Subjective Tones 507
4.11 Theories of Hearing 507

5. Applied Acoustics, Instruments and Apparatus

5.3 Acoustical Impedance Measurement (see also 11.4) 507
5.4 Analyzers and Filters. Acoustic Filters. Oscillographs (see also 5.15) 507
5.7 Frequency Standards. Frequency Measuring and Recording Instruments 508
Hearing Instruments (see 4.5 and 5.5)
5.8 Loudspeakers. Horns (see also 5.13 and 5.17) 508
5.9 Microphones, Vibration Microphones, and Microphone Calibration Equipment 508
5.11 Oscillators 508
5.13 Public Address Systems and Sound Picture Installations 508
5.15 Sound Level Meters. Level Recorders. Sound Pressure Measurement (see also 5.4) 508
5.16 Sound Recording and Reproducing Speech Instruments (see 9.6) 508
5.17 Telephone Receivers 509

6. Musical Instruments and Music

6.1 General, Unclassified 509
6.2 Bells 509
6.5 Organs (see also 6.9) 509
6.6 Pianos 509
6.7 Violin Family 509
6.9 Electronic Musical Instruments 509

7. Noise

7.1 General, Unclassified (see also 5.15) 509
7.7 Machinery Noise 509

9. Speech and Singing

9.1 General, Unclassified 510
9.2 Anatomy of Speech Organs 510
9.4 Frequency of Occurrence of Speech Sounds 510
9.5 Frequency Analyses of Vowels and Other Speech Sounds 510
9.6 Instruments Relating to Speech and Singing 510
9.7 Speech Pitch 510
9.8 Speech Power. Singing Voice Power 510

10. Supersonics (Ultrasonics)

10.1 General, Unclassified 510
10.4 Gases, Supersonic Velocities, Dispersion, and Absorption (see also 11.3) 510
10.5 Liquids, Supersonic Velocities, Dispersion, and Absorption 511
10.6 Solids, Supersonic Velocities, Dispersion, and Absorption 511
10.7 Physical Effects of Supersonic Waves 511

11. Waves and Vibrations

11.1 General, Unclassified 511
11.2 Diffraction and Interference 511
11.3 Propagation of Sound. Absorption during Propagation. Velocity of Sound (see also 10.4) 511
11.4 Radiation from Vibrating Objects. Acoustical Impedance. Mechanical Impedance (see also 5.3) 511
11.5 Sounds of Finite Amplitude 511
11.6 Sound Transmission Through Tubes 511
11.7 Theory of Waves and Vibrating Systems (see also 5.4) 511

12. General, Unclassified 511

Author Index to Volume 11

Numerals in bold face type refer to volume numbers. (A) designates abstracts; (T) designates title only; (L) designates Letter to the Editor.

Arts, Jan. "Bells." **11**: 485(L)—1940
—— Sound of bells. Jottings from my experiences with the sound of bells. The strike note. **11**: 321—1940

Bachem, A. Genesis of absolute pitch. **11**: 434—1940
Ballantine Stuart. Book review. **11**: 255—1939
Bartholomew, Wilmer T. Digest of modern voice research. **11**: 374(A)—1940
—— Paradox of voice teaching. **11**: 446—1940
Baumzweiger, Benjamin. Application of piezoelectric vibration pick-ups to measurement of acceleration, velocity and displacement. **11**: 303, 379(A)—1940
—— Graphical determination of the random efficiency of microphones. **11**: 477—1940
Beasley, W. C. Characteristics and distribution of deafness in the population of the United States. **11**: 166(A)—1939
—— Prevalence of nerve deafness in the population. **11**: 378(T)—1940
Beranek, Leo L. Absolute calibration of microphones (review). **11**: 153—1939
—— Acoustical properties of especially fine toned violins (review). **11**: 257—1939
—— Construction and properties of a variable acoustic resistance standard (review). **11**: 492—1940
—— Electromechanical feed-back system for vacuum-tube voltmeters with logarithmic indication (review). **11**: 366—1940
—— Inhomogeneity of the magnetic field of a dynamic loudspeaker (review). **11**: 261—1939
—— Normal modes and damping of gas bubbles in liquids (review). **11**: 366—1940
—— Normal modes of vibration of rooms with nonplanar walls and an investigation of diffuse sound reflection (review). **11**: 364—1940
—— On new results in research on violins (review). **11**: 490—1940
—— On sound propagation at grazing incidence to porous materials (review). **11**: 489—1940
—— On testing of metals by supersonics (review). **11**: 367—1940
—— On the perception of vibrations (review). **11**: 487—1940
—— Performance of rectangular rooms with one treated surface. **11**: 379(A)—1940
—— Piezoelectric measurements on the absolute auditory threshold for bone conduction (review). **11**: 259—1939
—— Sound Damping in Rigid and Elastic Wall Tubes (review). **11**: 154—1939
—— Supersonic phenomena (review). **11**: 262—1939
—— (see Hunt, F. V.) **11**: 80—1939
Bhatt, N. B. Effect of an absorbing wall on the decay of normal frequencies. **11**: 67, 160(A)—1939
Bilhuber, Paul H. and C. A. Johnson. Influence of the soundboard on piano tone quality. **11**: 166(A)—1939; **11**: 311—1940
Bolt, Richard H. Angular distribution of normal modes. **11**: 160(A)—1939
—— Normal modes of vibration in room acoustics: angular distribution theory. **11**: 74—1939
—— Normal modes of vibration in room acoustics: experimental investigations in nonrectangular enclosures. **11**: 184—1939
Bragg, Sir William. Combination tones in sound. **11**: 5—1939
—— Demonstration of combination tones. **11**: 162(A)—1939
—— Reminiscences in acoustics. **11**: 161(T)—1939
Bray, Charles W. (see Wever, Ernest Glen). **11**: 427—1940
Brillouin, Léon. Acoustical wave propagation in pipes. **11**: 10—1939
—— Discussion. **11**: 100—1939
Brittain, F. H. Loudness of continuous spectrum noise and its application to loudness measurements. **11**: 113—1939
Brooks, J. Woolson. Parasitic infestations in acoustical materials. **11**: 375(A)—1940
Brown, Spencer F. and Dorothy Oliver. Organic speech abnormalities associated with cleft palate. **11**: 376(T)—1940
Burris-Meyer, Harold. Sound in the theater. **11**: 346, 374(T), 380(A)—1940

Chrisler, V. L. Discussion. **11**: 96—1939
Coile, R. C. Development of modern microphones. **11**: 163(A)—1939
—— Parabolic sound concentrator. **11**: 167(A)—1939
Colwell, R. C. (see Stewart, J. K.) **11**: 147—1939
Corey, Victor B. and G. W. Stewart. Alteration in the liquid structure of electrolytes as indicated by acoustic waves. **11**: 378(A)—1940
Cowan, Milton. Studies of the vocal cords in relation to the quality of certain speech sounds. **11**: 376(T), 380(A)—1940
—— **and Paul Griffith.** Applications of the Buchmann-Meyer effect in the calibration of phonograph recording and reproducing equipment. **11**: 380(A)—1940

Darracott, H. T. (see Ham, L. B.) **11**: 333—1940
Dudley, Homer. Remaking speech. **11**: 165(A), 169—1939; **11**: 373(T)—1940
Dunn, H. K. and S. D. White. Statistical measurements on conversational speech. **11**: 278, 374(A)—1940

Edwards, Preston. Suggestion for simplified musical notation. **11**: 323—1940
Eyring, Carl F. Discussion. **11**: 104—1939

Fairbanks, Grant. Recent experimental investigations of vocal pitch in speech. **11**: 373(A), 457—1940
Field, George S. Discussion. **11**: 101—1939

Firestone, F. A. Artificial larynx for speaking and choral singing by one person. **11:** 357—1940
—— Demonstration of how one person can give a public speech or sing a quartet without using his vocal chords. **11:** 376(T)—1940
—— Current publications on acoustics. **11:** 153, 257—1939; **11:** 364, 487—1940
—— References to contemporary papers on acoustics. **11:** 156, 263—1939; **11:** 370, 494—1940
Fletcher, Harvey. Acoustical Society of America. Its aims and trends. **11:** 13, 161(T)—1939
—— Auditory patterns. A demonstration lecture. **11:** 163(A)—1939
Freehafer, J. E. Acoustical impedance of an infinite hyperbolic horn. **11:** 467—1940
Frick, R. H. and P. S. Veneklasen. Audiofrequency harmonic generator system. **11:** 166(A)—1939

Gardner, Mark B. (see Steinberg, John). **11:** 270—1940
Geiger, Paul H. Loudness level to loudness conversion chart. **11:** 308—1940
Glover, Ralph P. Review of cardioid type unidirectional microphones. **11:** 296, 379(A)—1940
Gray, Giles W. Minimum duration of perceptible speech sounds. **11:** 376(T)—1940
Griffith, Paul (see Cowan, Milton). **11:** 380(A)—1940

Hall, William M. Acoustic transmission line for impedance measurement. **11:** 140—1939
Ham, L. B. and H. T. Darracott. Vibratory characteristics of vibrafram. **11:** 333—1940
Harry, W. R. (see Marshall, R. N.) **11:** 164(A)—1939
Hasbrouck, H. J. Comments on paper by Professor F. V. Hunt presented at the meeting of the Acoustical Society of America at Iowa State University. **11:** 382—1940
Horner, George L. Practical application of acoustical correction in university buildings. **11:** 375(A)—1940
Hubbard, J. C. Book review. **11:** 255, 256—1939
Hunt, F. V. Absorption coefficient problem. **11:** 38—1939
—— Concluding remarks. **11:** 106—1939
—— Research on the absorption coefficient problem. **11:** 159(A)—1939
—— **and J. A. Pierce.** Stylus-groove relations and their influence on phonograph reproducer design. **11:** 379(A)—1940
——, **L. L. Beranek and D. Y. Maa.** Analysis of sound decay in rectangular rooms. **11:** 80—1939

Jacobs, C. R. Discussion. **11:** 102—1939
Johnson, C. A. (see Bilhuber, Paul H.) **11:** 166(A)—1939; **11:** 311—1940
Jones, Arthur Taber. Recent investigations of organ pipes. **11:** 122—1939
Jones, R. Clark. On the theory of fluctuations in the decay of sound. **11:** 324; Erratum. 485—1940
Josephson, E. M. Progressive deafness associated with disorders of vitamin A metabolism. **11:** 167(A)—1939

Kent, Earle L. New electronic musical instrument. **11:** 352—1940
Klapman, S. J. Interaction impedance of a system of circular pistons. **11:** 165(A)—1939; **11:** 289—1940
Knudsen, Vern O. Discussion. **11:** 105—1939
—— Ear to the future. **11:** 29, 162(A)—1939
—— Some practical aspects of architectural acoustics. **11:** 375(A), 383—1940
—— (see Watson, N. A.) **11:** 406—1940
Kock, Winston E. Analysis of transients. **11:** 163(A)—1939

Lawrence, Merle (see Wever, Ernest Glen) **11:** 427—1940
Leedy, H. A. Noise and vibration isolation. **11:** 164(A)—1939; **11:** 341—1940
Lewis, Don and Curtis Tuthill. Resonant frequencies and damping constants of resonators involved in the production of sustained vowels "O" and "Ah." **11:** 451—1940
Lifschitz, Samuel. Fluctuation of the hearing threshold. **11:** 118—1939
Lindahl, Robert and Hale J. Sabine. Measurement of impact sound transmission through floors. **11:** 379(A), 401—1940
Lindsay, R. B. Filtration of oblique elastic waves in stratified media. **11:** 165(A), 178—1939
Lloyd, Llewelyn S. Note on just intonation. **11:** 440—1940
Lurie, M. H. Studies of acquired and inherited deafness in animals. **11:** 377(A), 420—1940

Maa, D. Y. (see Hunt, F. V.) **11:** 80—1939
Marshall, R. N. and W. R. Harry. New microphone providing uniform directivity over an extended frequency range. **11:** 164(A)—1939
Martin, Daniel W. Book review. **11:** 255—1940
Massa, Frank. Microphone efficiency: a discussion and proposed definition. **11:** 164(A), 222—1939
Maxfield, J. P. (see Potwin, C. C.) **11:** 48, 162(A)—1939
—— **and C. C. Potwin.** Planning functionally for good acoustics. **11:** 375(A), 390—1940
Montgomery, H. C. Hearing survey at the New York and San Francisco World's Fairs. **11:** 378(A)—1940
Morrical, Keron C. Sound insulation characteristics for ideal partitions. **11:** 164(A), 211—1940
Morris, R. M. and G. M. Nixon. Discussion. **11:** 102—1939
Morse, Philip M. Some aspects of the theory of room acoustics. **11:** 56—1939
—— Transmission of sound inside pipes. **11:** 205—1939

Nixon, G. M. (see Morris, R. M.) **11:** 102—1939
Norris, R. F. Discussion. **11:** 103—1939

Oliver, Dorothy (see Brown, Spencer F.) **11:** 376(T)—1940

Parkinson, John S. Adaptation of acoustical materials to auxiliary functions. **11:** 375(A)—1940
—— Discussion. **11:** 97—1939
Pellam, John R. Sound diffraction and absorption by a strip of absorbing material. **11:** 396—1940
Pepinsky, Abe. Pitch as a function of the amount and rate of frequency modulation. **11:** 376(T), 381(A)—1940
Peterson, E. D. Nonlinear characteristics of cutting heads. **11:** 380(T)—1940

Phelps, W. D. Sound source for investigating microphone distortion. **11**: 164(A), 219—1940
Pierce, J. A. (see Hunt, F. V.) **11**: 379(A)—1940
Potwin, C. C. (see Maxfield, J. P.) **11**: 375(A), 390—1940
—— **and J. P. Maxfield.** Modern concept of acoustical design. **11**: 48, 162(A)—1939

Railston, William. Supersonic dispersion in vapors. **11**: 107—1939
Reger, Scott N. Correlations between air and bone conduction acuity measurements in different types of deafness. **11**: 378(A)—1940
Riffolt, N. (see Thompson, L.) **11**: 233—1939
Rogers, Robert. Attenuation of sound in ducts. **11**: 164(A)—1939
—— Attenuation of sound in tubes. **11**: 480—1940

Sabine, Hale J. What acoustical treatment will and will not do. **11**: 375(A)—1940
—— (see Lindahl, Robert) **11**: 379(A), 401—1940
Sabine, Paul E. Architectural acoustics: its past and its possibilities. **11**: 21, 161(A)—1939
—— Measurement of sound absorption coefficients from the viewpoint of the testing laboratory. **11**: 41—1939
—— Measurements in the reverberation chamber. **11**: 159(A)—1939
Saunders, F. A. Conclusions from further measurements on old and new violins. **11**: 165(A)—1939
—— Studies of old and new violins. **11**: 374(T)—1940
Schlenker, Vesper A. Architectural acoustic testing with electro-thermal noise as a test source. **11**: 164(A)—1939
Scott, H. H. Degenerative sound analyzer. **11**: 225—1939
Small, Arnold. Tone production on the stringed instrument. **11**: 374(T), 381(A)—1940
Snidecor, J. C. Comparative study of vocal pitch in impromptu speaking and oral reading of the same material. **11**: 376(T)—1940
Stanton, G. T. Correlation of sound absorption coefficients with field measurements. **11**: 45—1939
—— Measurements in the field. **11**: 160(A)—1939
Steinberg, John C. Speech production as revealed by vocal cord photography and sound wave analysis. **11**: 373(A)—1940
—— **and Mark B. Gardner.** On the auditory significance of the term hearing loss. **11**: 270—1940
Stewart, G. W.: Book review. **11**: 369—1940
—— (see Corey, Victor B.) **11**: 378(A)—1940
Stewart, J. K. and R. C. Colwell. Calculations of Chladni patterns. **11**: 147—1939
Stoddard, Dean George D. Tribute to Professor George Walter Stewart and Dean Carl Emil Seashore. **11**: 267—1940
Strother, Charles R. Use of an extrinsic laryngeal muscle as an abductor of the vocal cords. **11**: 376(T)—1940

Thompson, L. Propagation of shock waves in air. II. **11**: 245—1939
—— **and N. Riffolt.** Propagation of shock waves in air. I. **11**: 233—1939
Thomson, William T. Effect of rotatory and lateral inertia on flexural vibration of prismatic bars. **11**: 198—1939
Trimmer, John D. Resonant frequencies of certain pipe combinations. **11**: 129—1939
Tuthill, Curtis (see Lewis, Don) **11**: 451—1940

Veneklasen, P. S. (see Frick, R. H.) **11**: 166(A)—1939

Waterfall, Wallace. Events leading to the formation of the Acoustical Society of America. **11**: 11, 161(T)—1939
Watson, F. R. Discussion. **11**: 105—1939
—— Journal of the Acoustical Society of America. **11**: 15, 162(A)—1939
Watson, N. A. Selective amplification in hearing aids. **11**: 377(A)—1940
—— **and V. O. Knudsen.** Selective amplification in hearing aids. **11**: 406—1940
Wever, Ernest Glen, Charles W. Bray and Merle Lawrence. Locus of distortion in the ear. **11**: 427—1940
White, S. D. (see Dunn, H. K.) **11**: 278, 374(A)—1940
Williamson, Chas. Design for a keyboard instrument in just intonation. **11**: 216—1939
Willig, F. J. Possible variation of sound absorption coefficients with intensity. **11**: 95, 160(A)—1939

Young, Robert W. Book review. **11**: 369—1940
—— Terminology for logarithmic frequency units. **11**: 134, 166(A)—1939

Analytic Subject Index to Volume 11

Numerals in bold face type refer to volume numbers. (S) designates supplements; (T) designates title only; (L) designates Letter to the Editor

1. Acoustical Society of America

1.1 General, Unclassified

Acoustical Society news. Translation service. **11**:486—1940

Acoustical Society of America. Its aims and trends. Harvey Fletcher. **11**: 13—1939

Journal of the Acoustical Society of America. F. R. Watson. **11**: 15—1939

ABSTRACT

Journal of the Acoustical Society. F. R. Watson. **11**: 162—1939

1.2 Constitution and By-Laws

Constitution and by-laws. **11**: 171(S)—1939

1.3 History

Events leading to the formation of the Acoustical Society of America. Wallace Waterfall. **11**: 11—1939

History. **11**: 173(S)—1939

Past and present officers of Acoustical Society. **11**: 170(S)—1939

Past members of Executive Council. **11**: 170(S)—1939

SUBTITLES

History of the journal. F. R. Watson. **11**: 15—1939

Number of members, journal subscribers, and total expenditures each year. Harvey Fletcher. **11**: 14—1939

1.5 Meetings

Acoustical Society news. **11**: 152, 254—1939; 362, 486—1940

Program of the tenth anniversary meeting, New York, New York, May 15, 1939. **11**: 159—1939

Program of the twenty-second meeting, Iowa City, Iowa, November 2, 1939. **11**: 373—1940

Tenth anniversary meeting (general information). **11**: 1—1939

1.6 Members and Membership Lists

J. E. R. Constable, obituary notice. **11**: 152—1939

Edward Elway Free, obituary notice. **11**: 363—1940

Geographical index of members. **11**: 182(S)—1939

Membership list, July, 1939. **11**: 174(S)—1939

New members. **11**: 254—1939; 362, 486—1940

Photograph of Acoustical Society organization group, 1928. **11**: 2—1939

Photograph of G. R. Anderson. **11**: 4—1939

Photograph of banquet, tenth anniversary meeting. **11**: 3—1939

Photograph of Sir William Bragg. **11**: 4, 5—1939

Photograph of Léon Brillouin. **11**: 4—1939

Photograph of F. A. Firestone. **11**: 4, 16—1939

Photograph of Harvey Fletcher. **11**: 4, 13—1939

Photograph of E. E. Free. **11**: 1—1939

Photograph of V. O. Knudsen. **11**: 4, 29—1939

Photograph of J. P. Maxfield. **11**: 4—1939

Photograph of D. C. Miller. **11**: 4—1939

Photograph of Madeline M. Mitchell. **11**: 17—1939

Photograph of G. M. Nixon. **11**: 1—1939

Photograph of P. E. Sabine. **11**: 4, 21—1939

Photograph of F. A. Saunders. **11**: 4, 5—1939

Photograph of J. C. Steinberg. **11**: 1—1939

Photograph of Wallace Waterfall. **11**: 4, 11—1939

Photograph of F. R. Watson. **11**: 4, 5, 15—1939

Photograph of E. C. Wente. **11**: 18—1939

John Redfield, obituary notice. **11**: 254—1939

Tribute to Professor George Walter Stewart and Dean Carl Emil Seashore, November 4, 1939. George D. Stoddard. **11**: 267—1940

SUBTITLE

Number of members, journal subscribers, and total expenditures each year. Harvey Fletcher. **11**: 14—1939

1.7 Letters to the Editor

"Bells." Jan Arts. **11**: 485(L)—1940

2. Architectural Acoustics

2.1 General, Unclassified

PAPERS

Architectural acoustics: Its past and its possibilities. Paul E. Sabine. **11**: 21—1939

Some practical aspects of architectural acoustics. Vern O. Knudsen. **11**: 383—1940

Sound in the theater. Harold Burris-Meyer. **11**: 346—1940

ABSTRACT

Architectural acoustics, its past and its possibilities. P. E. Sabine. **11**: 161—1939

2.2 Auditorium Design

PAPERS

Modern concept of acoustical design. C. C. Potwin and J. P. Maxfield. **11**: 48—1939

Planning functionally for good acoustics. J. P. Maxfield and C. C. Potwin. **11**: 390—1940

Some practical aspects of architectural acoustics. Vern O. Knudsen. **11**: 383—1940

Sound in the theater. Harold Burris-Meyer. **11**: 346—1940

SUBTITLE

Suggestions on auditorium design. C. C. Potwin and J. P. Maxfield. **11**: 55—1939

ABSTRACTS

Modern concept of acoustical design. C. C. Potwin and J. P. Maxfield. **11**: 162—1939

Planning functionally for good acoustics. J. P. Maxfield and C. C. Potwin. **11**: 375—1940

Practical application of acoustical correction in university buildings. George L. Horner. **11**: 375—1940

Some practical aspects of architectural acoustics. Vern O. Knudsen. **11**: 375—1940

2.3 Studies of Existing Auditoria

SUBTITLES

Measured frequency reverberation characteristic, Temple of Religion, New York World's Fair. C. C. Potwin and J. P. Maxfield. **11**: 52—1939

Some typical errors in room acoustics. Vern O. Knudsen. **11**: 385—1940

Voder room. Bell System exhibit. C. C. Potwin and J. P. Maxfield. **11**: 53—1939

2.4 Sound Absorption, Theory and Methods of Measurement

PAPERS

Absorption coefficient problem. F. V. Hunt. **11**: 38—1939

Correlation of sound absorption coefficients with field measurements. G. T. Stanton. **11**: 45—1939

Discussion of absorption coefficient symposium. V. L. Chrisler **11**: 96, John S. Parkinson 97, L. Brillouin 100, George S. Field 101, R. M. Morris and G. M. Nixon 102, C. R. Jacobs 102, R. F. Norris 103, Carl F. Eyring 104, F. R. Watson 105, V. O. Knudsen 105, F. V. Hunt 106—1939

Measurement of absorption coefficients and the application of acoustical absorbents. A symposium. **11**: 37—1939

Measurement of sound absorption coefficients from the viewpoint of the testing laboratory. Paul E. Sabine. **11**: 41—1939

On sound propagation at grazing incidence to porous materials. K. Schuster (Akustische Zeits. **4**, 335 (1939)). **11**: 489—1940

Possible variation of sound absorption coefficients with intensity. F. J. Willig. **11**: 95—1939

Sound diffraction and absorption by a strip of absorbing material. John R. Pellam. **11**: 396—1940

Vibratory characteristics of vibrafram. L. B. Ham and H. T. Darracott. **11**: 333—1940

SUBTITLES

Absorbing material containing punched holes. John R. Pellam. **11**: 399—1940

Comparison of absorption coefficients as measured at the National Bureau of Standards and at Riverbank Laboratories. Paul E. Sabine. **11**: 42—1939

Normal impedance. Philip M. Morse. **11**: 56—1939

Variation in measured coefficient with area and magnitude of absorption coefficient. V. L. Chrisler. **11**: 97—1939

Variation of acoustic impedance ratio with frequency, for different positions of absorbing material in a room. Richard H. Bolt. **11**: 78—1939

ABSTRACTS

Adaptation of acoustical materials to auxiliary functions. John S. Parkinson. **11**: 375—1940

Architectural acoustic testing with electro-thermal noise as a test source. Vesper A. Schlenker. **11**: 164—1939

Comments on paper by Professor F. V. Hunt at the meeting of the Acoustical Society of America at Iowa State University. H. J. Hasbrouck. **11**: 382—1940

Measurements in the field. G. T. Stanton. **11**: 160—1939

Measurements in the reverberation chamber. P. E. Sabine. **11**: 159—1939

Parasitic infestations in acoustical materials. J. Woolson Brooks. **11**: 375—1940

Possible variation of sound absorption coefficients with intensity. F. J. Willig. **11**: 160—1939

Research on the absorption coefficient problem. F. V. Hunt. **11**: 159—1939

What acoustical treatment will and will not do. Hale J. Sabine. **11**: 375—1940

2.5 Measured Sound Absorption Coefficients

PAPER

Vibratory characteristics of vibrafram. L. B. Ham and H. T. Darracott. **11**: 333—1940

SUBTITLES

Celotex. F. V. Hunt, L. L. Beranek and D. Y. Maa. **11**: 87, 93—1939

Impedance of celotex tile board. N. B. Bhatt. **11**: 72—1939

Impedance of wool felt. N. B. Bhatt. **11**: 72—1939

2.7 Reverberation, Theory and Calculation of

PAPERS

Analysis of sound decay in rectangular rooms. F. V. Hunt, L. L. Beranek and D. Y. Maa. **11**: 80—1939

Effect of an absorbing wall on the decay of normal frequencies. N. B. Bhatt. **11**: 67—1939

Normal modes of vibration in room acoustics: Angular distribution theory. Richard H. Bolt. **11**: 74—1939

Normal modes of vibration in room acoustics: Experimental investigations in nonrectangular enclosures. Richard H. Bolt. **11**: 184—1939

On the theory of fluctuations in the decay of sound. R. Clark Jones. **11**: 324; Erratum. **11**: 485—1940

Some aspects of the theory of room acoustics. Philip M. Morse. **11**: 56—1939

SUBTITLES

Angular distribution of normal modes in a cube. Richard H. Bolt. **11**: 76—1939

Chart exhibiting solutions of the equation $\coth \pi(\alpha - j\beta) = (\gamma/\eta)e^{j\phi}(\beta - j\alpha)$. F. V. Hunt, L. L. Beranek and D. Y. Maa. **11**: 90—1939

Computation chart exhibiting contours of constant $\eta\delta_n$ and contours of constant ratio, $\mu = \delta_0/\delta_n$. F. V. Hunt, L. L. Beranek and D. Y. Maa. **11**: 91—1939

Computation of decay constants for individual modes of vibration. F. V. Hunt, L. L. Beranek and D. Y. Maa. **11**: 85—1939

High absorption at one wall, low absorption at remaining walls. F. V. Hunt. L. L. Beranek and D. Y. Maa. **11**: 82—1939

Normal frequency diagram of reverberation chamber. F. V. Hunt, L. L. Beranek and D. Y. Maa. **11**: 81—1939

Normal modes for rectangular rooms. Philip M. Morse. **11**: 207—1939

Normal modes of vibration of rooms with nonplanar walls and an investigation of diffuse sound reflection. E. Skudrzyk. **11**: 364—1940

Pressure amplitude of standing wave as function of distance between two parallel walls of a room. Philip M. Morse. **11**: 65—1939

Standing wave patterns in nonrectangular enclosures. Richard H. Bolt. **11**: 190—1939

Standing waves of sound in a room. Philip M. Morse. **11**: 58—1939

Variation of acoustic impedance ratio with frequency, for different positions of absorbing material in a room. Richard H. Bolt. **11**: 78—1939

ABSTRACTS

Angular distribution of normal modes. Richard H. Bolt. **11**: 160—1939

Effect of an absorbing wall on the decay of normal frequencies. N. B. Bhatt. **11**: 160—1939

Performance of rectangular rooms with one treated surface. Leo L. Beranek. **11**: 379—1940

2.9 Sound Transmission, Theory and Methods of Measurement

PAPERS

Measurement of impact sound transmission through floors. Robert Lindahl and Hale J. Sabine. **11**: 401—1940

Sound insulation characteristics for ideal partitions. Keron C. Morrical. **11**: 211—1939

SUBTITLES

Contours of transmission loss necessary to reduce symphony music to loudness level indicated. **11**: 213—1939

Contours of uniform loudness reduction for music. Keron C. Morrical. **11**: 212—1939

ABSTRACTS

Measurement of impact sound transmission through floors. Robert Lindahl and Hale J. Sabine. **11**: 379—1940

Sound insulation characteristics for ideal partitions. Keron C. Morrical. **11**: 164—1939

2.10 Measurements of Sound Transmission

SUBTITLES

Gypsum tile. Keron C. Morrical. **11**: 213—1939

Impact sound transmission of concrete, asphalt tile, asphalted felt, rubber tile, carpet, linoleum, celotex, cork tile and wood floor. Robert Lindahl and Hale J. Sabine. **11**: 403—1940

Pyrobar. Keron C. Morrical. **11**: 213—1939

2.11 Vibration Insulating Supports

PAPERS

Noise and vibration isolation. H. A. Leedy. **11**: 341—1940

On the perception of vibrations. George v. Békésy (Akustische Zeits. **4**, 316 (1939)). **11**: 487—1940

ABSTRACT

Noise and vibration isolation. H. A. Leedy. **11** :164—1939

3. Books and Bibliographies

3.1 Book Reviews

Applied acoustics. Harry F. Olson and Frank Massa. **11**: 369—1940

Complex variable and operational calculus. N. W. McLachlan. **11**: 255—1939

Musical slide-rule. Ll. S. Lloyd. **11**: 369—1940

Supersonics: Science of inaudible sound. R. W. Wood. **11**: 256—1939

Table relating frequency to cents. Robert W. Young. **11**: 255—1939

Ultrasonics and their scientific and technical applications. Ludwig Bergmann. **11**: 255—1939

3.2 Bibliographies

References to contemporary papers on acoustics. **11**: 156, 263—1939; 370, 494—1940

4. Ear and Hearing

4.1 General, Unclassified

PAPERS

Ear to the future. Vern O. Knudsen. **11**: 29—1939

On the perception of vibrations. George v. Békésy (Akustische Zeits. **4**, 316 (1939)). **11**: 487—1940

Selective amplification in hearing aids. N. A. Watson and V. O. Knudsen. **11**: 406—1940

SUBTITLE

Bibliography on hearing. F. H. Brittain. **11**: 116—1939

ABSTRACTS

Auditory patterns. Demonstration lecture. Harvey Fletcher. **11**: 163—1939

Ear to the future. V. O. Knudsen. **11**: 162—1939

4.2 Anatomy and Physiology of the Ear

SUBTITLE

Organs of Corti of various animals, magnified sections. M. H. Lurie. **11**: 421—1940

4.4 Deafness

PAPERS

On the auditory significance of the term hearing loss. John C. Steinberg and Mark B. Gardner. **11**: 270—1940

Selective amplification in hearing aids. N. A. Watson and V. O. Knudsen. **11**: 406—1940

Studies of acquired and inherited deafness in animals. M. H. Lurie. **11**: 420—1940

SUBTITLES

Amelioration of impaired hearing. Vern O. Knudsen. **11**: 29—1939

Average audiograms for groups reporting different degrees of hearing. John C. Steinberg and Mark B. Gardner. **11**: 275—1940

Form of articulation curves for normal and deafened observers. John C. Steinberg and Mark B. Gardner. **11**: 273—1940

Methods of prescribing selective amplification. N. A. Watson and V. O. Knudsen. **11**: 413—1940

Relation between hearing loss and the absolute sensitivity of the ear. John C. Steinberg and Mark B. Gardner. **11**: 276—1940

Relation between hearing loss and the capacity for interpreting everyday speech. John C. Steinberg and Mark B. Gardner. **11**: 275—1940

Relation between hearing loss and the interpretation of speech sounds. John C. Steinberg and Mark B. Gardner. **11**: 272—1940

Relation between hearing loss and the loss in loudness sensation. John C. Steinberg and Mark B. Gardner. **11**: 270—1940

Typical loudness loss for conductive deafness. John C. Steinberg and Mark B. Gardner. **11**: 270—1940

Typical loudness loss for nerve deafness. John C. Steinberg and Mark B. Gardner. **11**: 271—1940

ABSTRACTS

Characteristics and distribution of deafness in the population of the United States. W. C. Beasley. **11**: 166—1939

Correlations between air and bone conduction acuity measurements in different types of deafness. Scott N. Reger. **11**: 378—1940

Hearing survey at the New York and San Francisco World's Fairs. H. C. Montgomery. **11**: 378—1940

Prevalence of nerve deafness in the population. Willis C. Beasley. **11**: 378(T)—1940

Progressive deafness associated with disorders of vitamin A metabolism. E. M. Josephson. **11**: 167—1939

Studies of acquired and inherited deafness in animals. M. H. Lurie. **11**: 377—1940

4.5 Instruments Relating to Hearing

PAPER

Selective amplification in hearing aids. N. A. Watson and V. O. Knudsen. **11**: 406—1940

SUBTITLES

Amelioration of impaired hearing. Vern O. Knudsen. **11**: 29—1939

Articulation tests of hearing aids. N. A. Watson and V. O. Knudsen. **11**: 408—1940

Ear defenders. Vern O. Knudsen. **11**: 33—1939

Methods of prescribing selective amplification. N. A. Watson and V. O. Knudsen. **11**: 413—1940

ABSTRACT

Selective amplification in hearing aids. N. A. Watson. **11**: 377—1940

4.6 Loudness. Threshold Determinations

PAPERS

Fluctuation of the hearing threshold. Samuel Lifshitz. **11**: 118—1939

Loudness level to loudness conversion chart. Paul H. Geiger. **11**: 308—1940

Loudness of continuous spectrum noise and its application to loudness measurements. F. H. Brittain. **11**: 113—1939

SUBTITLES

Bibliography on loudness. F. H. Brittain. **11**: 116—1939

Bibliography on noise measurement. F. H. Brittain. **11**: 117—1939

Piezoelectric measurements on the absolute auditory threshold for bone conduction. G. v. Békésy. **11**: 259—1939

Relation between hearing loss and the absolute sensitivity of the ear. John C. Steinberg and Mark B. Gardner. **11**: 276—1940

Relation between hearing loss and the loss in loudness sensation. John C. Steinberg and Mark B. Gardner. **11**: 270—1940

Relation between intensity of noise and intensity of 1000~ note judged to be equally loud. F. H. Brittain. **11**: 113—1939

ABSTRACT

Hearing survey at the New York and San Francisco World's Fairs. H. C. Montgomery. **11**: 378—1940

4.7 Masking

SUBTITLE

Typical articulation results for an ear masked by noise. John C. Steinberg and Mark B. Gardner. **11**: 274—1940

4.9 Pitch

PAPERS

Genesis of absolute pitch. A. Bachem. **11**: 434—1940

Terminology for logarithmic frequency units. Robert W. Young. **11**: 134—1939

4.10 Subjective Tones

PAPERS

Combination tones in sound. Sir William Bragg. **11**: 5—1939

Locus of distortion in the ear. Ernest Glen Wever, Charles W. Bray and Merle Lawrence. **11**: 427—1940

ABSTRACT

Demonstration of combination tones. Sir William Bragg. **11**: 162—1939

4.11 Theories of Hearing

ABSTRACT

Auditory patterns. Demonstration lecture. Harvey Fletcher. **11**: 163—1939

5. Applied Acoustics. Instruments and Apparatus

5.3 Acoustical Impedance Measurement (see also 11.4)

PAPERS

Acoustic transmission line for impedance measurement. William M. Hall. **11**: 140—1939

Construction and properties of a variable acoustic resistance standard. K. Schuster and W. Stohr (Akustische Zeits. **4**, 253 (1939)). **11**: 492—1940

SUBTITLES

Acoustical impedance measurement bibliography. William M. Hall. **11**: 149—1939

Apparatus for measuring mechanical impedance. George v. Békésy (Akustische Zeits. **4**, 316 (1939)). **11**: 487—1940

5.4 Analyzers and Filters. Acoustic Filters. Oscillographs (see also 5.15)

PAPER

Degenerative sound analyzer. H. H. Scott. **11**: 225—1939

SUBTITLE
Octave filter oscillograph. Vern O. Knudsen. **11**: 34—1939

5.7 Frequency Standards. Frequency Measuring and Recording Instruments

PAPER
Terminology for logarithmic frequency units. Robert W. Young. **11**: 134—1939

SUBTITLE
Frequency level as a semitone count. Robert W. Young. **11**: 137—1939

ABSTRACTS
Table relating frequency to cents. Robert W. Young. Book. **11**: 255—1939
Terminology for logarithmic frequency units. R. W. Young. **11**: 166—1939

5.8 Loudspeakers, Horns (see also 5.13 and 5.17)

PAPERS
Acoustical impedance of an infinite hyperbolic horn. J. E. Freehafer. **11**: 467—1940
Interaction impedance of a system of circular pistons. S. J. Klapman. **11**: 289—1940

SUBTITLE
Inhomogeneity of the magnetic field of a dynamic loudspeaker. W. Reinhard. **11**: 261—1939

ABSTRACT
Interaction impedance of a system of circular pistons. S. J. Klapman. **11**: 165—1939

5.9 Microphones, Vibration Microphones, and Microphone Calibration Equipment

PAPERS
Application of piezoelectric vibration pick-ups to measurement of acceleration, velocity and displacement. Benjamin Baumzweiger. **11**: 303—1940
Graphical determination of the random efficiency of microphones. Benjamin Baumzweiger. **11**: 477—1940
Microphone efficiency: Discussion and proposed definition. Frank Massa. **11**: 222—1939
Review of cardioid type unidirectional microphones. Ralph P. Glover. **11**: 296—1940
Sound source for investigating microphone distortion. William D. Phelps. **11**: 219—1939

SUBTITLES
Absolute calibration of microphones. W. Ernsthausen. (Akustische Zeits. **4**, 13 (1939)) **11**: 153—1939
Frequency characteristic crystal vibration pick-up. Benjamin Baumzweiger. **11**: 305—1940
Harmonic distortion in an RCA-44A microphone. William D. Phelps. **11**: 221—1939
Random energy response. Ralph P. Glover. **11**: 301—1940
Vibration pick-up calibration. Benjamin Baumzweiger. **11**: 304—1940

ABSTRACTS
Application of piezoelectric vibration pick-ups to measurement of acceleration, velocity and displacement. Benjamin Baumzweiger. **11**: 379—1940
Development of modern microphones. R. C. Coile. **11**: 163—1939
Microphone efficiency: Discussion and proposed definition. Frank Massa. **11**: 164—1939
New microphone providing uniform directivity over an extended frequency range. R. N. Marshall and W. R. Harry. **11**: 164—1939
Parabolic sound concentrator. R. C. Coile. **11**: 167—1939
Review of cardioid type unidirectional microphones. Ralph P. Glover. **11**: 379—1940
Sound source for investigating microphone distortion. W. D. Phelps. **11**: 164—1939

5.11 Oscillators

ABSTRACT
Audiofrequency harmonic generator system. R. H. Frick and P. S. Veneklasen. **11**: 166—1939

5.13 Public Address Systems and Sound Picture Installations

PAPER
Sound in the theater. Harold Burris-Meyer. **11**: 356—1940

ABSTRACT
Sound in the theater. Harold Burris-Meyer. **11**: 374(T), 380—1940

5.15 Sound Level Meters. Level Recorders. Sound Pressure Measurement (see also 5.4)

PAPERS
Degenerative sound analyzer. H. H. Scott. **11**: 225—1939
Loudness level to loudness conversion chart. Paul H. Geiger. **11**: 308—1940

SUBTITLES
Electromechanical feed-back system for vacuum-tube voltmeters with logarithmic indication. Ludwig Keidel. (Akustische Zeits. **4**, 169 (1939)) **11**: 366—1940
Logarithmic vacuum-tube voltmeter. H. H. Scott. **11**: 230—1939

5.16 Sound Recording and Reproducing (see also 5.13)

SUBTITLE
Notes on sound recording. H. J. Hasbrouck. **11**: 382—1940

ABSTRACTS
Applications of the Buchmann-Meyer effect in the calibration of phonograph recording and reproducing equipment. Milton Cowan and Paul Griffith. **11**: 380—1940
Nonlinear characteristics of cutting heads. E. D. Peterson. **11**: 380(T)—1940
Stylus-groove relations and their influence on phonograph reproducer design. F. V. Hunt and J. A. Pierce. **11**: 379—1940

5.17 Telephone Receivers

PAPER

Remaking speech. Homer Dudley. **11:** 169—1939

ABSTRACT

Remaking speech. Homer Dudley. **11:** 165—1939; 373(T)—1940

6. Musical Instruments and Music

6.1 General, Unclassified

PAPERS

Artificial larynx for speaking and choral singing by one person. F. A. Firestone. **11:** 357—1940

Design for a keyboard instrument in just intonation. Chas. Williamson. **11:** 216—1939

Note on just intonation. Llewelyn S. Lloyd. **11:** 440—1940

Suggestion for simplified musical notation. Preston Edwards. **11:** 323—1940

Terminology for logarithmic frequency units. Robert W. Young. **11:** 134—1939

SUBTITLES

Frequency level as a semitone count. Robert W. Young. **11:** 137—1939

Frequency level subscript notation for musical tones. Robert W. Young. **11:** 136—1939

Modern acoustics and music. Vern O. Knudsen. **11:** 34—1939

Musical slide-rule. Ll. S. Lloyd. Book. **11:** 369—1940

ABSTRACTS

Table relating frequency to cents. Robert W. Young. Book. **11:** 255—1939

Terminology for logarithmic frequency units. R. W. Young. **11:** 166—1939

6.2 Bells

PAPER

Sound of bells. Jottings from my experiences with the sound of bells. Strike note. Jan Arts. **11:** 321—1940

6.5 Organs (see also 6.9)

PAPERS

Design for a keyboard instrument in just intonation. Chas. Williamson. **11:** 216—1939

Recent investigations of organ pipes. Arthur Taber Jones. **11:** 122—1939

SUBTITLE

Spectra of different organ pipes. Arthur Taber Jones. **11:** 123—1939

6.6 Pianos

PAPERS

Design for a keyboard instrument in just intonation. Chas. Williamson. **11:** 216—1939

Influence of the soundboard on piano tone quality. Paul H. Bilhuber and C. A. Johnson. **11:** 311—1940

SUBTITLES

Maps showing distribution of vibrational energy in soundboard. Paul H. Bilhuber and C. A. Johnson. **11:** 313—1940

Motor-driven striker for producing succession of strikes of equal intensity. Paul H. Bilhuber and C. A. Johnson. **11:** 312—1940

Slope level for measuring angle that the strings make with the horizontal plane at the point where they cross the bridge. Paul H. Bilhuber and C. A. Johnson. **11:** 315—1940

ABSTRACT

Influence of the soundboard on piano tone quality. Paul H. Bilhuber and C. A. Johnson. **11:** 166—1939

6.7 Violin Family

PAPER

On new results in research on violins. H. Backhaus and G. Weymann (Akustische Zeits. **4**, 302 (1939)). **11:** 490—1940

SUBTITLE

Acoustical properties of especially fine toned violins. H. Meinel. **11:** 257—1939

ABSTRACTS

Conclusions from further measurements on old and new violins. F. A. Saunders. **11:** 165—1939

Studies of old and new violins. Frederick A. Saunders. **11:** 374(T)—1940

Tone production on the stringed instrument. Arnold Small. **11:** 374(T), 381—1940

6.9 Electronic Musical Instruments

PAPERS

Artificial larynx for speaking and choral singing by one person. F. A. Firestone. **11:** 357—1940

New electronic musical instrument. Earle L. Kent. **11:** 352—1940

ABSTRACT

Audiofrequency harmonic generator system. R. H. Frick and P. S. Veneklasen. **11:** 166—1939

7. Noise

7.1 General, Unclassified (see also 5.15)

PAPERS

Loudness level to loudness conversion chart. Paul H. Geiger. **11:** 308—1940

Loudness of continuous spectrum noise and its application to loudness measurements. F. H. Brittain. **11:** 113—1939

Noise and vibration isolation. H. A. Leedy. **11:** 341—1940

SUBTITLES

Bibliography on noise measurement. F. H. Brittain. **11:** 117—1939

Conquest of noise. Vern O. Knudsen. **11:** 32—1939

Ear defenders. Vern O. Knudsen. **11:** 33—1939

7.7 Machinery Noise

SUBTITLES

Classification of noises. H. H. Scott. **11:** 225—1939

Noise analysis of a domestic refrigerator. H. H. Scott. **11:** 231—1939

ABSTRACT

Noise and vibration isolation. H. A. Leedy. **11:** 164—1939

9. Speech and Singing

9.1 General, Unclassified

PAPERS

Artificial larynx for speaking and choral singing by one person. F. A. Firestone. **11**: 357—1940

Paradox of voice teaching. Wilmer T. Bartholomew. **11**: 446—1940

Sound in the theater. Harold Burris-Meyer. **11**: 346—1940

Statistical measurements on conversational speech. H. K. Dunn and S. D. White. **11**: 278—1940

SUBTITLE

Emotional content of speech versus speech pitch. Grant Fairbanks. **11**: 459—1940

ABSTRACTS

Demonstration of how one person can give a public speech or sing a quartet without using his vocal cords. F. A. Firestone. **11**: 376(T)—1940

Digest of modern voice research. Wilmer T. Bartholomew. **11**: 374—1940

Minimum duration of perceptible speech sounds. Giles W. Gray. **11**: 376(T)—1940

Organic speech abnormalities associated with cleft palate. Spencer F. Brown and Dorothy Oliver. **11**: 376(T)—1940

Speech production as revealed by vocal cord photography and sound wave analysis. John C. Steinberg. **11**: 373—1940

Studies of the vocal cords in relation to the quality of certain speech sounds. Milton Cowan. **11**: 376(T), 380—1940

Use of an extrinsic laryngeal muscle as an abductor of the vocal cords. Charles R. Strother. **11**: 376(T)—1940

9.2 Anatomy of Speech Organs

ABSTRACT

Speech production as revealed by vocal cord photography and sound wave analysis. John C. Steinberg. **11**: 373—1940

9.4 Frequency of Occurrence of Speech Sounds

SUBTITLE

Emotional content of speech *versus* speech pitch. Grant Fairbanks. **11**: 459—1940

9.5 Frequency Analyses of Vowels and Other Speech Sounds

PAPER

Resonant frequencies and damping constants of resonators involved in the production of sustained vowels "O" and "Ah." Don Lewis and Curtis Tuthill. **11**: 451—1940

9.6 Instruments Relating to Speech and Singing

PAPERS

Artificial larynx for speaking and choral singing by one person. F. A. Firestone. **11**: 357—1940

Remaking speech. Homer Dudley. **11**: 169—1939

ABSTRACTS

Remaking speech. Homer Dudley. **11**: 165—1939

Sound in the theater. Harold Burris-Meyer. **11**: 380—1940

9.7 Speech Pitch

PAPER

Recent experimental investigations of vocal pitch in speech. Grant Fairbanks. **11**: 457—1940

ABSTRACTS

Comparative study of vocal pitch in impromptu speaking and oral reading of the same material. J. C. Snidecor. **11**: 376(T)—1940

Pitch as a function of the amount and rate of frequency modulation. Abe Pepinsky. **11**: 376(T), 381—1940

Recent studies of fundamental vocal pitch in speech. Grant Fairbanks. **11**: 373—1940

9.8 Speech Power. Singing Voice Power

PAPER

Statistical measurements on conversational speech. H. K. Dunn and S. D. White. **11**: 278—1940

SUBTITLES

Long interval r.m.s. pressures. H. K. Dunn and S. D. White. **11**: 282—1940

Ratios of peak to r.m.s. pressures in speech. H. K. Dunn and S. D. White. **11**: 281—1940

R.m.s. pressures in one-eighth-second intervals. H. K. Dunn and S. D. White. **11**: 279—1940

Spectral distribution of peak acoustic powers in speech and symphony music. Keron C. Morrical. **11**: 212—1939

Spectrum of average total speech power. H. K. Dunn and S. D. White. **11**: 284—1940

Speech peak pressures on one-eighth-second intervals. H. K. Dunn and S. D. White. **11**: 278—1940

ABSTRACT

Statistical measurements on conversational speech. H. K. Dunn and S. D. White. **11**: 374—1940

10. Supersonics (Ultrasonics)

10.1 General, Unclassified

SUBTITLE

Supersonic phenomena. William T. Richards. **11**: 262—1939

ABSTRACTS

Supersonics: science of inaudible sounds. R. W. Wood. Book. **11**: 256—1939

Ultrasonics and their scientific and technical applications. Ludwig Bergmann. Book. **11**: 255—1939

10.4 Gases, Supersonic Velocities, Dispersion, and Absorption (see also 11.3)

PAPER

Supersonic dispersion in vapors. William Railston. **11**: 107—1939

SUBTITLES

Absorption and velocity for the following vapors: carbon disulphide, benzene, chloroform, methylene chloride, ethyl

alcohol, acetone, ethyl iodide, carbon tetrachloride, ethyl ether, methyl iodide, methyl alcohol. William Railston. **11**: 110—1939

Normal modes and damping of gas bubbles in liquids. E. Meyer and K. Tamm. **11**: 366—1940

Supersonic interferometer for vapors. William Railston. **11**: 108—1939

10.5 Liquids, Supersonic Velocities, Dispersion, and Absorption

ABSTRACT

Alteration in the liquid structure of electrolytes as indicated by acoustic waves. Victor B. Corey and G. W. Stewart. **11**: 378—1940

10.6 Solids, Supersonic Velocities, Dispersion, and Absorption

SUBTITLE

On testing of metals by supersonics. Fritz Kruse. **11**: 367—1940

10.7 Physical Effects of Supersonic Waves

SUBTITLE

On testing of metals by supersonics. Fritz Kruse. **11**: 367—1940

11. Waves and Vibrations

11.1 General, Unclassified

PAPER

On the perception of vibrations. George v. Békésy (Akustische Zeits. **4**, 316 (1939)). **11**: 487—1940

11.2 Diffraction and Interference

PAPER

Sound diffraction and absorption by a strip of absorbing material. John R. Pellam. **11**: 396—1940

11.3 Propagation of Sound. Absorption During Propagation. Velocity of Sound

SUBTITLE

Normal modes and damping of gas bubbles in liquids. E. Meyer and K. Tamm. **11**: 366—1940

11.4 Radiation from Vibrating Objects. Acoustical Impedance. Mechanical Impedance (see also 5.3)

PAPERS

Construction and properties of a variable acoustic resistance standard. K. Schuster and W. Stöhr (Akustische Zeits. **4**, 263 (1939)). **11**: 492—1940

Interaction impedance of a system of circular pistons. S. J. Klapman. **11**: 289—1940

SUBTITLE

Apparatus for measuring mechanical impedance. George v. Békésy (Akustische Zeits. **4**, 316 (1939)). **11**: 487—1940

ABSTRACT

Interaction impedance of a system of circular pistons. S. J. Klapman. **11**: 165—1939

11.5 Sounds of Finite Amplitude

PAPERS

Propagation of shock waves in air. I. L. Thompson and N. Riffolt. **11**: 233—1939

Propagation of shock waves in air. II. L. Thompson. **11**: 245—1939

SUBTITLE

Supersonic phenomena. William T. Richards. **11**: 262—1939

11.6 Sound Transmission Through Tubes

PAPERS

Acoustical wave propagation in pipes. Léon Brillouin. **11**: 10—1939

Attenuation of sound in tubes. R. Rogers. **11**: 480—1940

Resonant frequencies of certain pipe combinations. John D. Trimmer. **11**: 129—1939

Transmission of sound inside pipes. Philip M. Morse. **11**: 205—1939

SUBTITLES

Sound damping in rigid and elastic wall tubes. E. Waetzmann and W. Wenke. (Akustische Zeits. **4**, 1 (1939)). **11**: 154—1939

Tubes of rectangular section. Philip M. Morse. **11**: 205—1939

ABSTRACT

Attenuation of sound in ducts. Robert Rogers. **11**: 164—1939

11.7 Theory of Waves and Vibrating Systems (see also 5.4)

PAPERS

Calculation of Chladni patterns. J. K. Stewart and R. C. Colwell. **11**: 147—1939

Combination tones in sound. Sir William Bragg. **11**: 5—1939

Effect of rotatory and lateral inertia on flexural vibration of prismatic bars. William T. Thomson. **11**: 198—1939

Filtration of oblique elastic waves in stratified media. R. B. Lindsay. **11**: 178—1939

SUBTITLES

Normal modes and damping of gas bubbles in liquids. E. Meyer and K. Tamm. **11**: 366—1940

Photographs of Chladni patterns. J. K. Stewart and R. C. Colwell. **11**: 149—1939

ABSTRACTS

Analysis of transients. Winston E. Kock. **11**: 163—1939

Complex variable and operational calculus. N. W. McLachlan. Book. **11**: 255—1939

Demonstration of combination tones. Sir William Bragg. **11**: 162—1939

Filtration of oblique elastic waves in stratified fluid-solid media. R. B. Lindsay. **11**: 165—1939

12. General, Unclassified

References to contemporary papers on acoustics. **11**: 156, 263—1939; 370, 494—1940